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June 2003

An Integrated Computer-Controlled System for Marine Mammal Auditory Testing

Hearing Test Program (HTP.EXE) User's Guide

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EXECUTIVE SUMMARY

Public concern over the potential effects of anthropogenic (human-generated) noise on marine mammal behavior and hearing has been increasing. One of the first steps in evaluating the potential effects of underwater sound on marine mammals is to establish the hearing abilities of these animals.

This document describes the operation of the computer program HTP.EXE, which is designed to allow computer control of marine mammal hearing tests. The system uses the psychophysical "staircase" method and a behavioral response paradigm to estimate hearing thresholds in trained subjects. The program is suitable for isolated hearing threshold measurements and paired measurements made before and after exposure to threshold-affecting treatments (e.g., exposure to intense sound or other ototoxic agent). Although specifically designed for use with marine mammals, the program is appropriate for use with any subject trained for participation in a behavioral response paradigm.

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ADMINISTRATIVE INFORMATION

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1. INTRODUCTION

1.1 OVERVIEW

This document describes the operation of the computer program HTP.EXE, which is designed to allow computer control of marine mammal hearing tests. The system uses psychophysical "staircase" method and a behavioral response paradigm to estimate hearing thresholds in trained subjects. The program is suitable for isolated hearing threshold measurements and paired measurements made before and after exposure to some threshold-affecting treatment (e.g., exposure to intense sound or other ototoxic agent). Although specifically designed for use with marine mammals, the program is appropriate for use with any subject trained for participation in a behavioral response paradigm.

1.2 SOFTWARE REQUIREMENTS

1.2.1 Operating System

The HTP.EXE program will work with Microsoft® Windows 95, 98, 2000, or XP.

1.2.2 MATLAB® Command Window

The program requires the MATLAB® programming environment to perform some mathematical operations and to print results. When the HTP.EXE program is opened, the MATLAB® Command Window will open (minimized). The MATLAB® Command Window icon should appear on the Windows taskbar (see Figure 1) and remain there as long as the HTP.EXE program is running. If the MATLAB® Command Window is accidentally closed, the HTP.EXE program must be exited and run again.

Figure 1. HTP and MATLAB® Command Window taskbar icons.

1.3 HARDWARE REQUIREMENTS

At least one National Instruments "E-series" multifunction board is required. For full functionality, two E-series boards are required. The program was written specifically for use with the PCI-MIO-16E-1 board. Other boards may be used (provided they feature the DAQ-STC with two general purpose counters), but the user should confirm that any DAQ input limits specified are actually supported by the board. If the user selects an input range not supported by the board, the next lowest supported mode will be used, with no warning or error message. The program was developed for use with boards with 12-bit analog-to-digital conversion; use of a board that features 16-bit inputs/outputs has not been tested and may produce erratic results.

1.4 NOMENCLATURE

In this document, bold text indicates a specific menu item, front panel control or indicator, variable name, or a specific title. Menu items followed by an ellipsis (...) either contain additional front panels, screens, or dialogs.

1.5 SI UNITS

The International System of Units, universally abbreviated SI (from the French *Le Système International d'Unités*), is the modern metric system of measurement. The SI was established in 1960 and has become the dominant language of international commerce and trade (Taylor, 1995). The SI is founded on seven *SI base units* for seven mutually independent *base quantities*. The three most significant base quantities for acoustics are length, mass, and time, with base units of the meter (m), the kilogram, (kg), and the second (s), respectively.

Other quantities, called *derived quantities*, are defined in terms of the seven base quantities. An example of a derived quantity is the SI unit for density; density is mass per unit volume, or kilograms per cubic meter (kg/m^3). For convenience and ease of understanding, some SI-derived units have been given special names and symbols. For example, the SI unit of force is the newton (N): 1 N = 1 kg·m/s^2. The SI unit of pressure is the pascal (Pa): 1 Pa = 1 N/m^2.

To form decimal multiples and submultiples of SI units, there are 20 SI prefixes. Eight of the most common SI prefixes are listed in Table 1. Prefixes are applied as multiples of the SI unit, for example, 1000 meters = 1 kilometer (km). Note that, although a base unit, the kilogram contains a prefix.

Table 1. Common SI prefixes.

Multiple	Name	Symbol	Multiple	Name	Symbol
10^9	giga	G	0.01	centi	c
10^6	mega	M	0.001	milli	m
1 000	kilo	k	10^{-6}	micro	µ
0.1	deci	d	10^{-9}	nano	n

1.6 SOUND PRESSURE LEVEL AND THE DECIBEL

Sound pressure is defined as the small variation in the static pressure within a medium as a sound wave travels through it. The term *sound pressure level* (SPL) describes a sound pressure expressed in *decibels*. A decibel is 1/10 of a bel, a logarithmic unit named after Alexander Graham Bell. The logarithm of a physical unit is undefined (you can only compute the logarithm of a dimensionless number); therefore, to take the logarithm of a quantity with physical units (e.g., pressure), you must divide the quantity by a reference quantity with the same units. The equation for SPL is

$$SPL = 20\log_{10}\left(\frac{P}{P_{\text{ref}}}\right), \tag{1}$$

where P is the measured pressure and P_{ref} is some reference pressure. The standard reference for underwater sound is 1 µPa (1×10^{-6} Pa). The standard reference for airborne sound is 20 µPa (20×10^{-6} Pa). Because the reference quantities for underwater and airborne sound are different, airborne and underwater SPLs are not directly comparable. In addition, the loudness of sounds is a subjective attribute. An underwater sound and an airborne sound with the same SPL do not have the same loudness. The accepted terminology for specifying decibel quantities is to present the numeric value followed by the text "re" and the numeric value and units of the reference quantity. For example, a pressure of 1 Pa, expressed in decibels with a reference of 1 µPa, would be written 120 dB re 1 µPa.

The properties of logarithms and the form of equation (1) produce some useful results:

1. $\log_{10}(xy) = \log_{10} x + \log_{10} y$ and $\log_{10}(x/y) = \log_{10} x - \log_{10} y$

2. If $SPL_1 = 20\log_{10}\left(\frac{P_1}{P_{\text{ref}}}\right)$ and $SPL_2 = 20\log_{10}\left(\frac{P_2}{P_{\text{ref}}}\right)$, then $SPL_1 - SPL_2 = 20\log_{10}\left(\frac{P_1}{P_2}\right)$
3. $\log_{10}(2) \approx 0.3$, so if the sound pressure is doubled, the SPL increases by 6 dB.
4. $\log_{10}(3.16) \approx 0.5$, so if the sound pressure increases by a factor of 3.16, the SPL increases by 10 dB, which explains the use of the sequence 0.1, 0.316, 1.0, 3.16, 10, 31.6, etc. on the B&K 2635 charge amps; each setting increases the amplification by 10 dB.

2. INSTALLED COMPONENTS

2.1 HTP FOLDER

The program and data files are located in the HTP folder (Figure 2). Within this directory are 21 separate folders containing the source code, data files, and executable file for the HTP.EXE program, as well as data files created by the program. The HTP folder is referred to as the Base Path in the HTP.EXE program and may be changed by the user if necessary.

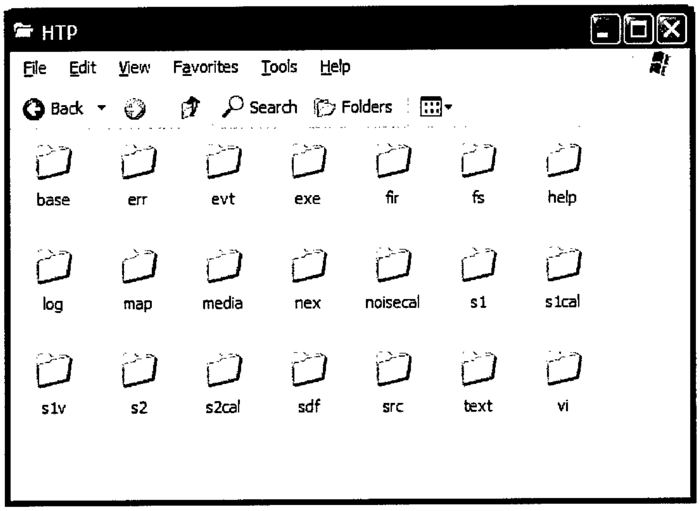


Figure 2. HTP folder contents.

2.2 SUB-FOLDERS

Each of the sub-folders within the HTP folder contains a specific file type. Table 2 contains a brief description of each sub-folder's contents.

Table 2. HTP sub-folder contents.

Sub-folder	Contents
base	Baseline hearing threshold data established for each subject. This folder should contain a separate text file for each subject listing the average baseline threshold at each frequency.
err	Contains a record of any errors encountered during program execution.
evt	Contains the event files (with the extension .EVT). These text files are created during a session and contain a record of the time each s1 and s2 tone were generated.
exe	Contains the executable file HTP.EXE (and other files needed for execution).
fir	Contains the noise specification files. These text files with the .FIR extension are created during masking noise calibration.
fs	Fatiguing stimulus (FS) files. These are single channel, 16-bit, signed, Motorola format binary files that contain the FS waveform.
help	Contains help documents in PDF and Microsoft® Word format.

Table 2. HTP sub-folder contents. (continued)

Sub-folder	Contents
log	Datalog folder. The datalog file is a database of test parameters and results.
map	Contains data files created when mapping the acoustic field.
media	Contains WAV files for sound effects.
nex	Setup files for the NEXUS conditioning amplifier (if present).
noisecal	Noise calibration results. These data files are created automatically when noise is saved during calibration. The files have the extension .MAT; these files are MATLAB® binary data files.
s1	Contains hydrophone recordings made during S1 tone presentations. Files are saved as two-channel, interleaved, 16-bit, signed integers, Intel format.
s1cal	S1 calibration results files. These are MATLAB® binary files with the extension .MAT.
s1v	S1 source files. These text files have the extension .s1v and contain the frequency, amplitude, rise/fall time, and delay for the S1 tones.
s2	Contains hydrophone recordings made during S2 tone presentations. Files are saved as single channel, 16-bit, signed integers, Intel format.
s2cal	S2 calibration results files. These files are created automatically during S2 calibration. The files have the extension .MAT; these files are MATLAB® binary data files.
sdf	Session data files. These text files contain the results of the test session; e.g., for each trial, tone amplitude, if the subject responded or not, etc.
src	S2 source files. These text files specify the S2 voltage necessary to generate each desired SPL.
text	Files specifying the text for pull-down menu controls (e.g., Trainer and Subject).
vi	HTP.EXE source code files.

2.3 PROGRAM CONVENTIONS

2.3.1 Closing Windows

Most windows appearing within the HTP.EXE program may not be closed using the Windows "Close Box" located in the upper right corner. This feature is disabled (the close box appears dimmed) because the HTP.EXE program often requires certain "housekeeping" functions to be performed after the user closes the window. Clicking the Windows close box simply "destroys" the window, without executing these additional functions. Some windows allow keyboard combinations such as Ctrl-W or Ctrl-X to close the window, in addition to the File menu Close or Exit options.

2.3.2 Front Panels, Controls, and Indicators

The term Front Panel refers to any HTP.EXE program window. Controls are items on a front panel that the user manipulates; for example, boxes to enter text or numbers, pull-down menus, buttons, switches, etc. Indicators are front panel items that display measured values or other data that the user does not normally change or interact with. Indicators may be number or text boxes, graphs, gauges, lights, etc. Sometimes a front panel object may be used as a control and an indicator (i.e., it displays measured values but the user may click on it and change the value).

2.3.3 Entering Numbers and Text

Keyboard characters (i.e., numbers or letters) entered into front panel controls or graph axes are not accepted by the program until either the mouse is clicked outside the text region or the Enter or Return key is pressed. When entering values into controls, the cursor will change into a flashing vertical bar. Once the program accepts the value, the cursor returns to the "hand" symbol.

2.3.4 Tip Strips

Many front panel controls and indicators have "tip strips" associated with them. Tip strips are brief text messages that appear when the mouse cursor is held over an object. To check for help on a front panel object, hold the mouse cursor over the object to see if a tip strip appears.

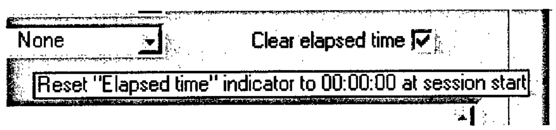


Figure 3. Tip strip example.

2.3.5 Working with Graphs

The user may manipulate the appearance of graphs by changing the axes limits or display mode. Many graph indicators possess groups of buttons, called **palettes**, which the user may use to change various graph attributes. Figure 4 shows the palettes used to change the graph properties and axes limits. The top group of three buttons is called the graph palette, which manipulates cursors, zoom in/out on a graph, or drag data within a graph window. The lower sets of buttons (the scale palettes) are used to change the x- and y-axes scale limits, autoscale, formatting, and plotting method.

Error! Reference source not found. shows a detail of the graph palette. Pressing one of these three buttons will change the appearance of the cursor (while over the graph window) and determine what happens when the user clicks or click-drags within the graph window. Only one button may be pressed at a time; the "active" button is shown in a darker color (the center button in Figure 4). The left button changes the cursor to the crosshairs symbol. With the crosshairs, the user may click-drag to move any cursors on the graph. The right button changes the cursor to a hand symbol; using the hand to click-drag within the graph window drags the data—it allows the user to view other areas of the graph.

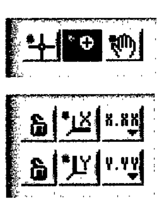


Figure 4. Graph and Scale palettes.

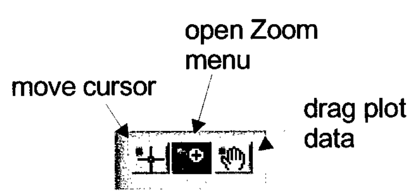


Figure 5. Graph palette detail.

The center button opens a separate dialog window (Figure 6) with several zoom options. The top row of buttons control the method of zooming in. From left to right, these three buttons allow the user to left-click and drag within the graph to zoom-in on a **rectangular region**, **horizontal region**, or **vertical region**. The lower left button will "un-do" the last zoom action. The lower center and right buttons zoom in and out, respectively, on any point left-clicked in the graph window.

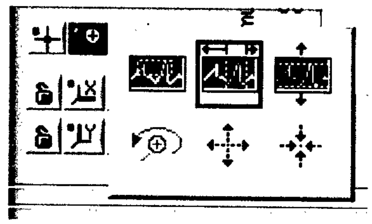


Figure 6. Graph palette zoom controls.

Figure 7 shows the x- and y-axis scale palettes. These buttons control various attributes of the x- and y-axes. The center button will perform an **autoscale** once. An autoscale sets the graph x- and y-axes limits to correspond to the maximum values in the data being plotted. Performing an autoscale is a good way to quickly see all of the data or recover from multiple or accidental zooming-in. Pressing the center button will autoscale one time only. If the data values change after this autoscale, the axes limits will not change to reflect this change. The left button toggles the autoscale lock. If the autoscale is locked, then an autoscale will be performed every time the data are plotted, keeping the entire data set visible despite changes in the numeric values. The autoscale for the x- and y-axes are independent; often, it is desirable to autoscale only the y-axis, for example, to keep measured amplitudes visible despite large changes, while keeping the x-axis autoscale off.

The right button opens a secondary menu (Figure 8) to control more detailed axis options. The **Format** menu item allows the user to change the format of the scale numbers, e.g., decimal or scientific notation. The **Precision** menu option controls the number of decimal digits displayed in the scale numbers. The **Mapping Mode** option determines whether the axis is plotted in linear or logarithmic mode.

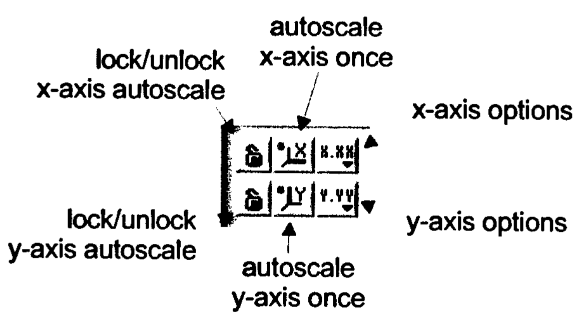


Figure 7. x- and y-axes scale palettes.

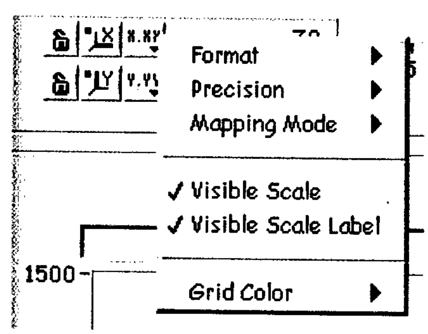


Figure 8. Additional axes options.

In addition to the graph and scale palettes, the user may also directly change the axes limits by first clicking on a numeric value on one of the axes scales and replacing the existing value.

Note that most graphs will use default values for the axes limits, format, precision, and mapping mode—user changes to these items will not be saved once the program is exited. Some graphs will record user changes to the axes limits (the main HTP.EXE staircase plot, for example).

3. HARDWARE SETUP

Figure 9 is a schematic of the hardware setup. The three main signal paths are (1) sounds generated by the computer and sent to the sound projectors, (2) sounds in the water picked up by the hydrophones and sent to the computer, and (3) video signals.

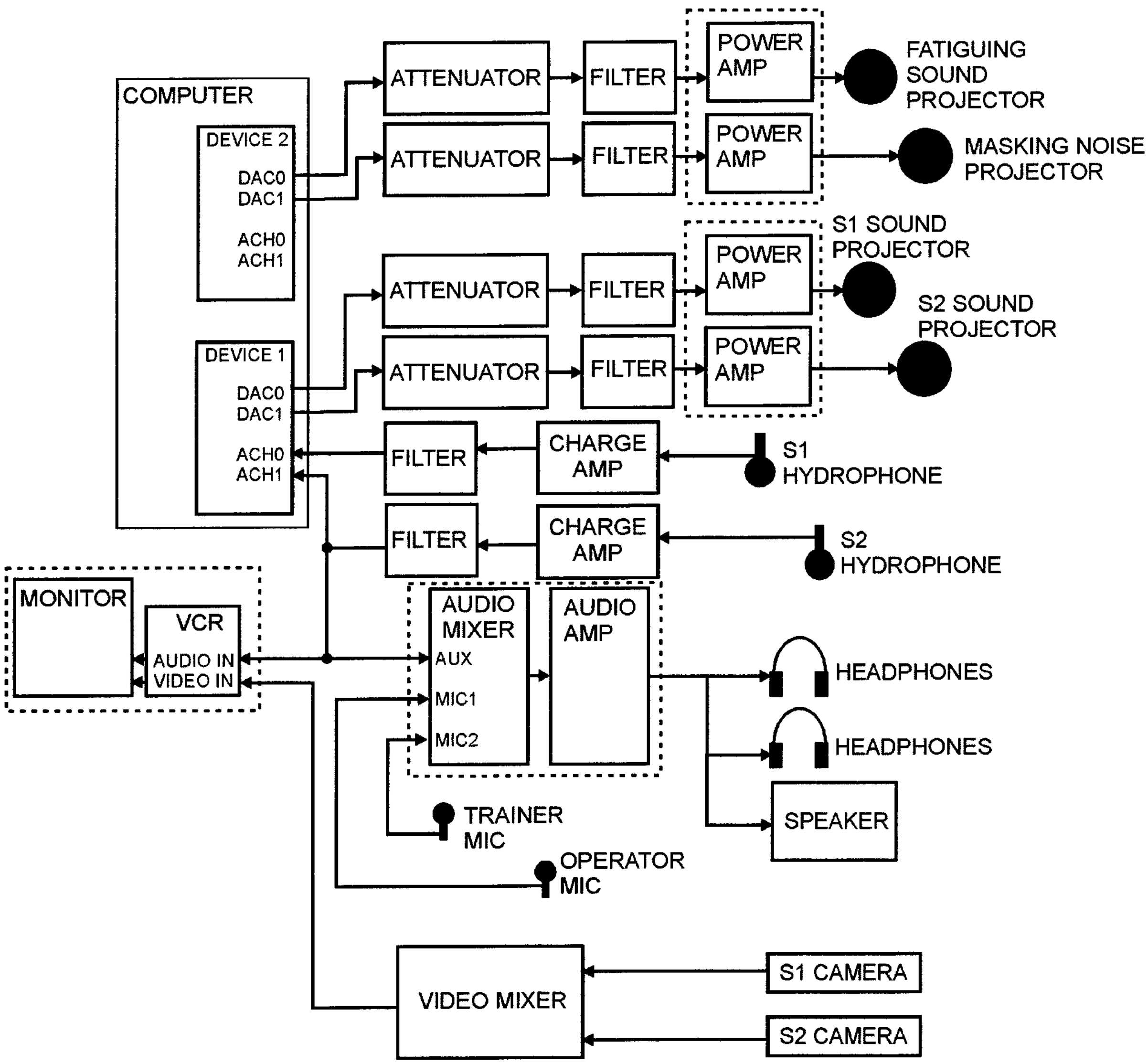


Figure 9. Hardware setup for marine mammal hearing tests.

The heart of the sound system is a personal computer (PC) with one or more multifunction data acquisition (DAQ) “boards” (the specific models used are National Instruments PCI-MIO-16E-1). Each DAQ board is assigned a unique *device number*. The computer uses the device number to identify a particular DAQ board (very important if more than one board is in the PC). Device 1 is normally used to generate and record the S1 and S2 signals. Device 2 (if present), is normally used to generate the fatiguing stimulus and masking noise.

Each DAQ board connects internally to one of the computer's PCI expansion slots. A 68-pin cable connects the back of the DAQ board (i.e., the part that sticks out from the PC) to a "BNC breakout box" with multiple BNC-type connectors on it. The breakout box allows the various inputs and outputs from the DAQ board to be easily connected to other hardware such as hydrophones and sound projectors.

3.1 SOUND GENERATION

The computer generates four types of sounds: (1) "S1 signals," or "start signals," used to cue the subject to the hearing test station; (2) "S2 tones," or hearing test tones; (3) a fatiguing stimulus (the "loud" sound that may cause a temporary threshold shift, or TTS); (4) masking noise, sometimes used to investigate the detection of sounds within noise or provide a "floor" effect in the presence of varying ambient noise.

Sound generation by the computer relies on *digital-to-analog conversion* (DAC). Sound waveforms are created within the computer as sequences of numbers, then converted to analog (electrical) signals by generating specific voltages at discrete time values. The generated voltages are updated at a speed specified by the DAC Update Rate. In this way, time-varying voltages are generated; the frequency of the analog signal generated depends on the update rate.

Each DAQ board may generate two signals simultaneously, that is, the board has two output *channels*. The channel 0 analog output signal appears at the DAC0 connector on the breakout box; the channel 1 output signal appears at DAC1. Within the HTP, the acronym DAC is always used to indicate what channel is used for sounds coming out of the computer. DAC0 is normally used for the S1 signal; DAC1 is normally used for the S2 signals. If a second device is present, it will possess its own DAC0 (normally used for the fatiguing stimulus) and DAC1 (normally used for masking noise).

3.1.1 Attenuators

The electrical signals generated by the computer are restricted to a voltage range of ±10 V. It is advantageous to operate near the upper end of this range (and the sound quality will degrade significantly if the DAC voltages are below approximately 0.05 to 0.1 V); however, 10 V is above the maximum allowable input for most electronic gear. The DAC signals coming out of the PC are therefore *attenuated* by first passing the signals through an electronic *attenuator*. The attenuator reduces the voltage coming out of the PC so that it may be input to other electronic equipment (such as amplifiers). The attenuator setting is in decibels; a decibel is a logarithmic unit indicating the ratio between two quantities. For an attenuator, the dB setting indicates how much the output signal is lowered relative to the input; therefore, ***increasing*** the attenuator setting ***lowers*** the SPL produced.

3.1.2 Filters

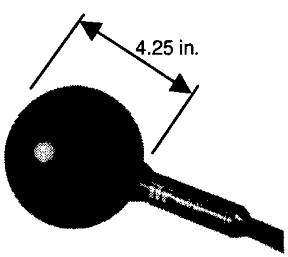
After being attenuated, the DAC signals are *filtered*. A *filter* is a device that selectively attenuates certain frequencies. Filters are generally one of three types: (1) a *high-pass* filter attenuates low frequencies and allows the high frequencies to "pass" through, (2) a *low-pass* filter attenuates high frequencies, and (3) a *band-pass* filter attenuates everything below one frequency and everything above a second frequency and allows frequencies in between to pass unaffected. The DAC process has an unavoidable side-effect of generating noise at a frequency corresponding to the DAC update rate (i.e., the rate at which the digital values are written to the DAC output channel). Therefore, DAC signals must be low-pass filtered at one-half the update rate, or lower, to remove high-frequency noise that occurs. The settings on the filter (what frequencies to pass) may depend on the particular test frequency; it may be necessary to change filter settings when testing widely separated frequencies.

3.1.3 Amplifiers

After filtering, the signals are amplified. The type of amplifiers (similar to home stereo amplifiers) are called *power amplifiers* because they are used to generate relatively high electric current and/or voltages necessary to "drive" the underwater sound sources.

3.1.4 Underwater Sound Projectors

After amplification, each signal is input to an *underwater sound projector*. An underwater sound projector is a *transducer*—a device that converts energy from one form to another. Underwater sound projectors convert electrical energy to acoustic energy (sound waves). Many sound projectors are made from *piezoelectric* ceramics (Figure 10). A piezoelectric material changes shape when an electric charge is applied to it. When an electrical voltage is applied to a piezoelectric sphere, for example, the sphere changes shape (vibrates) and generates sound in the water. To provide good coupling between the projector surface and the water and to prevent air bubbles from forming, the projector surface should be coated with a soap/water solution before immersion. If possible, each projector should be rinsed with fresh water after use. The projector surface should not be exposed to direct sunlight.



(A)

(B)

Figure 10. Two piezoelectric sound projector models: (A) ITC-1001 and (B) ITC-1042 (Not to scale).

3.2 SOUND RECORDING

Sounds at the S1 and S2 stations are recorded by the PC and stored on the hard disk. Sound recording by the computer relies on *analog-to-digital conversion* (ADC), the opposite of DAC. During ADC, an analog voltage is converted into a digital sequence of numbers; each number represents the value of the analog signal at some discrete time. The time interval between successive values is determined by the **Scan rate**. The scan rate tells the PC how many times per second to measure the value of the analog signal and convert it into a digital number. Higher frequency sounds require higher scan rates to adequately convert them to digital signals (to "digitize" them).

Each DAQ board may record up to eight signals simultaneously; however, only two are normally used. The individual input channels are referred to as ACH0, ACH1, .., ACH7. Within the HTP program, the acronym ACH is always used to indicate what channel is used for sounds coming into the computer. ACH0 is normally used for the hydrophone located at the S1 station; ACH1 is normally used for the hydrophone located at the S2 station.

To actually record the underwater sounds, they must first be converted to electrical signals, amplified, and filtered.

3.2.1 Hydrophones

Sounds in the water are measured using underwater microphones called *hydrophones*. Like sound projectors, hydrophones are electro-acoustic transducers, only they work in reverse: hydrophones convert underwater sound waves into electrical signals. Most hydrophones are made from piezo-electric ceramics shaped into spheres or cylinders (Figure 11). Particular care must be used in handling not only the hydrophone itself, but also the cable and connector. The cables should not be tightly coiled or bent at sharp angles. Hydrophones should be rinsed with fresh water after use and stored out of direct sunlight.

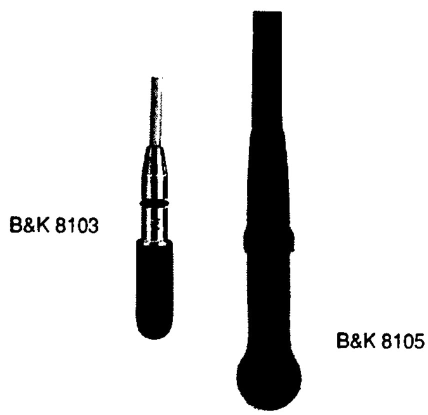


Figure 11. Piezoelectric hydrophone models B&K 8103 (left) and B&K 8105 (right).

3.2.2 Charge Amplifiers

The voltages generated by the hydrophones are typically very small, so they require amplification before they may be input to other devices (for filtering or storage). The hydrophone signals are input to *voltage amplifiers* or *charge amplifiers*. Charge amplifiers (charge amps) are recommended because for a piezoelectric hydrophone, it is the electric charge, not the voltage, that is proportional to the sound pressure. Two popular charge amps are the (1) B&K 2635, and (2) B&K NEXUS. The 2635 is a small, single-channel amplifier. The NEXUS is larger and has two channels. The charge amps should be enclosed in rugged, watertight cases with bulkhead BNC connectors for the input (the hydrophone), the output (to the equipment hut, normally), and an electrical ground (normally the water). The charge amps are battery-powered; batteries must be periodically re-charged. Both types of charge amps include adjustable band-pass filters.

The result of the hydrophone/charge amp combination is a voltage output (in millivolts) that is proportional to the sound pressure (in pascals). The sensitivity of the hydrophone/charge amp is specified in mV/Pa, the number of millivolts generated for a specified pressure in pascals. This quantity appears in several places in the HTP application, normally referred to as the **scale factor (mV/Pa)** or **hydrophone sensitivity (mV/Pa)**. The scale factor, K, is thus used to relate the sound pressure, P, and the voltage, E: $E = K \cdot P$.

3.2.2.1 Determining the Scale Factor (B&K 2635)

The scale factor for the B&K 2635 charge amps is calculated from the equation

$$K = \frac{S_{\mathrm{H}}\, V_{\mathrm{U}}}{S_{\mathrm{CA}}}, \tag{2}$$

where S_{H} is the hydrophone charge sensitivity in pC/Pa (obtained from the hydrophone calibration chart), S_{CA} is the charge amp sensitivity in pC/unit (indicated by the charge amp Transducer Sens. switches and decimal point indicator), and V_{U} is the charge amp mV/Unit Out setting. (The abbreviation pC stands for picocoulombs: pico (p) is the SI prefix for 10^{-12}, coulomb (C) is the unit for electric charge.) The value for S_{CA} is determined by the numerical value of the three Transducer Sens. dials and the position of the Transducer Sens. toggle switch. For example, if the dials are set to 123 and the switch is in the 0.1–1 position, then $S_{\mathrm{CA}} = 0.123$; if the switch is set to the 1–11 position, then $S_{\mathrm{CA}} = 1.23$. In normal circumstances, the charge amp Transducer Sens. S_{CA} is set equal to the hydrophone sensitivity, S_{H}, so that equation (2) simplifies to $K = V_{\mathrm{U}}$. Unfortunately, the B&K 8103 hydrophones have charge sensitivities around 0.09 pC/Pa, which is below the minimum value of 0.1 that the B&K 2635 may be set. In this case, it is customary to set S_{CA} to $10\times S_{\mathrm{H}}$ (e.g., use $S_{\mathrm{CA}} = 0.9$ if $S_{\mathrm{H}} = 0.09$), thus K will be equal to $V_{\mathrm{U}}/10$.

3.2.2.2 Determining the Scale Factor (B&K NEXUS)

The scale factor in mV/Pa may be directly read from the NEXUS charge amp display. If the scale factor is not visible, go to the Peak Meter display by selecting ≡⁝ from the NEXUS main menu.

3.2.2.3 Determining the Scale Factor (Voltage Amplifier)

Voltage amplifiers often specify the scale factor in units of dB re 1 V/μPa. To use the HTP, the scale factor must be converted to units of mV/Pa using $K = 10^{(9+K'/20)}$, where K is the scale factor in mV/Pa and K' is the scale factor in dB re 1 V/μPa.

3.2.3 Filters

The output from the charge amp is input to a band-pass filter. This filter is used for two purposes: (1) to remove low-frequency sounds produced by water currents and motion of the hydrophone itself, and (2) to remove high-frequency signals. Every signal to be digitized must be low-pass filtered to prevent high-frequency signals from creating errors in the digitized data. The low-pass filter must be set to a frequency less than one-half the scan rate, or lower.

3.3 WAVEFORM GENERATION AND TIMING

Accurate assessment of the subject's response latency, defined as the time interval from the stimulus onset to the response onset, requires synchronization between the sound generation and recording procedures. This synchronization is accomplished within the HTP by using one of the DAQ-STC digital counters to generate a waveform for use as a common timing reference, so that the sound generation and recording processes start at exactly the same time.

Figure 12 illustrates the relationships between the digital timing signal, sound generation, and recording events. The output signal from the DAQ-STC "general-purpose counter zero" (GPCTR0) is used as the main timing reference. The GPCTR0 output signal is a +5-V digital pulse with a width equal to the desired delay between sound recording and waveform generation. Data acquisition (i.e., hydrophone recording) is started or "triggered" when the GPCTR0 signal rises. The duration of the recording, the analog input (AI) duration, is set by the user. Sound generation (S1 and S2 only —see Section 4.1) is triggered when the GPCTR0 signal falls. Masking noise generation is triggered by the rise of the GPCTR0 signal. The fatiguing sound is triggered through software, rather than a hardware timing signal. The duration and rise/fall times of the sound waveforms are specified by the user. During S1 and S2 tone generation, the output from GPCTR1 is also triggered by the fall of the GPCTR0 signal. The width of the GPCTR1 pulse is equal to the sound duration. GPCTR1 output may be used to trigger an external function generator to produce audible "beeps" during sound generation. This procedure is essential during hearing tests at ultrasonic frequencies, so that the computer operator and the trainer know when the tones are presented to the subject. The S1, S2, FS, and masking noise start times are "locked" to the start of the hydrophone recording within approximately 30 µs. There is substantially more time fluctuation with the GPCTR1 signal, whose onset may vary up to ±5 ms relative to the start of the recording.

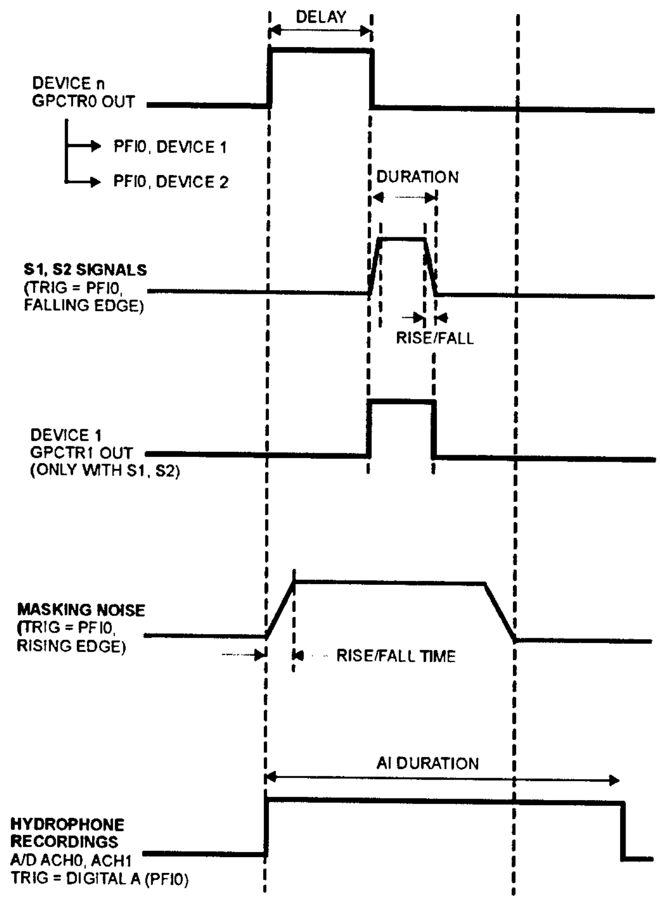


Figure 12. Relationships between input, output, and timing signals.

Some external connections are necessary to obtain proper connectivity between the various digital timing signals. In particular, programmable function input (PFI) connection zero must be connected to the output of the GPCTR0. This jumper between GPCTR0 and PFI0 is required (on both DAQ devices, if present) for normal operation of the HTP. Figure 13 illustrates the required external connections between GPCTR0 and devices 1 and 2.

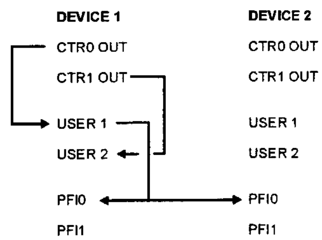


Figure 13. External DAQ connections required for HTP.

3.4 HEADSETS

In addition to going into the computer for digitization, the S2 hydrophone signal is also sent to an audio mixer (or amplifier with multiple audio inputs). Here, the hydrophone signal is combined (mixed) with the audio from the trainer and computer operator microphones. The mixed audio is amplified (using a conventional stereo amp) and sent to the headset speakers. The relative balance between the trainer and operator vocals and the hydrophone sound is obtained by adjusting the MIC1 (operator), MIC2 (trainer), and AUX (hydrophone) knobs or faders. The overall volume in each headphone is adjusted with the volume knob.

3.5 VIDEO

Video cameras are mounted on the S1 and S2 stations to give the operator a view of the subject during the tests. The video output from each camera is sent to a video mixer. The video mixer combines the two separate video signals into a single video signal with a split screen, showing each biteplate. The output from the video mixer goes to a VCR and to a video monitor. The VCR audio input is obtained from the S2 hydrophone signal (the AUX signal).

4. RUNNING A SESSION

4.1 SESSION OVERVIEW

The test apparatus consists of one or two underwater listening stations (Figure 14). In TTS testing, two stations, designated the "S1 station" and the "S2 station," are used to spatially separate the location of a potentially fatiguing sound exposure from the location of the hearing tests. The S1 station is the site for the presentation of a "start" signal to begin the hearing test, as well as the fatiguing stimulus. The actual hearing tests are conducted at the S2 station. In applications involving only hearing tests with no fatiguing sound exposure, only the S2 station is used.

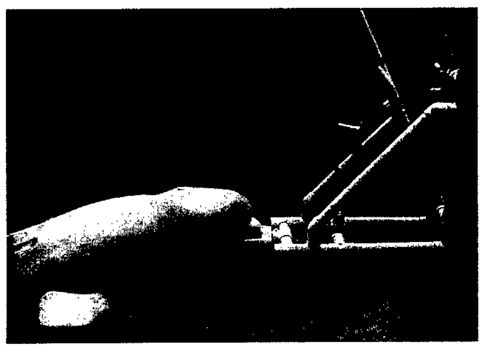


Figure 14. A white whale positioned on an underwater listening station.

During the hearing test procedure, each subject is presented with a number of hearing test tones (S2 tones) during a relatively long observation period, designated here as a "dive" (multiple dives are required to allow the subjects to periodically surface for air). Each dive begins with the trainer directing the animal (with a hand signal) to the S1 station. The subject is trained to remain on the S1 station until presented with the S1 start signal. Upon hearing the S1 start signal, the subject proceeds directly to the S2 station for hearing tests. If the S1 station is not used, the trainer cues the subject directly to the S2 station. Once the subject is positioned at the S2 station, S2 tones are presented. Each tone presentation is called a "trial." In addition to S2 tone trials, no-tone or "catch" trials may also be presented to the subject. The subject's responses to the catch trials is used to assess the subject's response bias (see below). The time interval between trials is often randomized.

Subjects are trained to produce an audible response if they hear a tone and to remain quiet otherwise. The amplitudes of the S2 tones are adjusted using a modified up/down **staircase procedure** (see Figure 15). The amplitude is decreased a certain amount following each hit (a response to a tone) and increased a certain amount following each miss (no response to a tone). After a variable number of tones, the trainer sounds an underwater buzzer that signals the animal to leave the S2 station and return to the surface for fish reward. The next dive then begins, if necessary.

There are two possible outcomes for each type of trial: a hit (a vocal response to a tone), a miss (no response to a tone), a correct rejection (CR) (no response to a catch trial), or a false alarm (FA) (a response to a catch trial). Note that hits and CRs are both correct responses and misses and FAs are both incorrect responses. The number of FAs is used to assess the subject's response bias, i.e., how liberal or conservative the subject was during the session. A relatively small number (e.g., in the 10 to 20% range) is perfectly acceptable. A subject that rarely commits FAs is likely to be very conservative (i.e., only responds when he is absolutely sure that he hears the sound) and therefore thresholds measured from this subject may be relatively high.

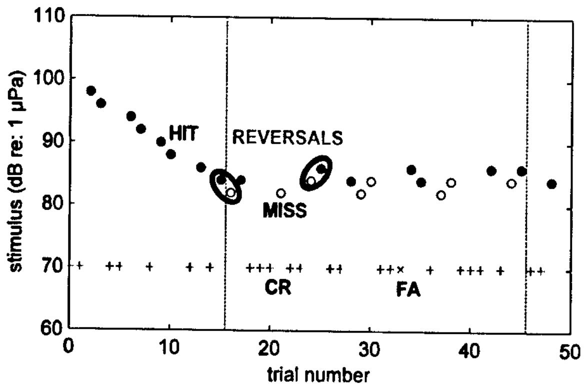


Figure 15. Example data from staircase procedure.

More detailed descriptions of the hearing test procedure may be found in Schlundt *et al.* (2000), and Finneran, Carder, and Ridgway (2002), and Finneran *et al.* (2000, 2002a, 2002b).

4.2 QUICK START

To start a hearing test, select **Begin New Session** from the **Session** menu. The Session Configuration dialog will appear. Each tab is used to set different options for the session. From the **Session** tab, change the **Animal code**, **Operator**, and **Trainer** parameters to the appropriate values. Use the **Methodology** tab to set the minimum and maximum interstimulus interval (ISI) and the catch trial percentage. Use the **S2** tab to select the appropriate S2 source file (created during calibration) and set the proper **Starting SPL** for the staircase type(s) being used. Press **OK**.

The front panel text message window will say "Waiting for Start Trigger". When the subject is positioned on the S1 biteplate, press the S1 button on the control box. This will send an S1 signal. If the front panel **Fatiguing Stim** button is lit, the fatiguing sound will be sent as well. After hearing the S1 signal, the subject will proceed to the S2 station. When the subject is positioned on the S2 biteplate, flip the control box S2 toggle switch to the UP position. A single S2 hearing test tone will be produced and the S2 hydrophone recording will be displayed in the lower window.

Following the S2 tone presentation and hydrophone recording, the text window will say "Waiting for response". Press the YES button on the control box if the subject responded. Press NO if the subject did not respond. Press the IGNORE button to disregard the trial (e.g., subject left the station). Press the EARLY button before keying in the response if the subject responded before the tone was presented or outside of a trial (not visible on the screen).

After the response button is pressed, the result of the trial will be displayed in the upper plot (exception: ignore trials are not displayed). The next trial will automatically begin if the S2 toggle switch is in the UP position when the response button was pressed. If the S2 switch is DOWN before the response button is pressed, the dive will end and the text window will say "Waiting for Start Trigger".

At the conclusion of the dive, the front panel Score indicator will update to show the score. The front panel text message window will display "Waiting for Start Trigger".

4.3 USING THE CONTROL BOX

The computer operator uses a control box (Figure 16) to interact with the computer. The box has six buttons and one toggle switch. The leftmost red button, the **S1 trigger**, sends the S1 signal. The button to the right and below the S1, the **on-station** button, is used to indicate when the subject is on and off the S2 station. The toggle switch, or **S2 trigger**, initiates hearing test trials. The three response buttons are used to record the subject's response during the trial: **yes**, meaning a vocal response; **no**, meaning no vocal response; **ignore**, ignore this trial (e.g., the subject left the S2 station). The last button, **early**, records a vocal response that occurs before or after a trial, that is, any vocal response that does not appear on-screen in the response window, or that appears, but began before the S2 tone.

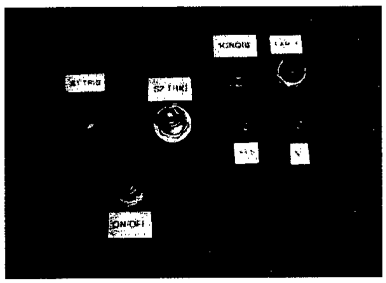


Figure 16. HTP control box.

Pressing the **S1 trigger** will cause a single S1 signal to be sent. This button may be pressed as many times in a row as necessary.

Pressing the **on-station** button toggles the front panel **Station** (OFF/ON) indicator.

The **S2 trigger** is a two-position toggle switch. S2 tones will be sent from the PC as long as the switch remains in the "UP" position. S2 tones may be sent without sending an S1 signal first. After each S2 is sent, the PC waits for the operator to press one of the three response buttons (**yes**, **no**, **ignore**).

Pressing one of the three response buttons will end the current trial and record the trial data to the PC hard disk. If the **S2 trigger** is UP, the next trial will begin immediately—there is no way to stop this next trial. Therefore, it is important that the operator flip OFF the **S2 trigger** before pressing the response button if the dive has ended; otherwise, another trial will begin. A yes response is defined as a response that begins in the window, after the red vertical bar (the red bar appears 50 ms after the tone start—it is impossible for the subjects to respond any faster than that). A response that begins before the red bar does not count. In this case, there is no response in the window, thus the correct response is No.

Pressing the **early** button will update a numeric indicator that keeps track of the number of "early" responses (Note, "early" here really means "out of the response window." Although these are "false positives," they are technically not "false alarms," which are defined as a yes response during a catch trial, as opposed to a yes response that occurs outside of a trial.)

Some important things to remember about the control box operation:

- Pressing a response button immediately begins the next trial.

- The computer will wait for the operator to press one of the three response buttons before proceeding, so do not feel like you have to hurry to key in the response.
- Do not worry about whether the trial was a tone trial or a catch trial—just key in whether the subject responded or not. The computer knows if the trial was a tone or a catch and will determine if it was a hit, miss, CR, or FA.
- If the trial was the last in a dive, the S2 toggle switch must be put into the DOWN position before the response button is pressed; otherwise, the next trial will begin automatically.
- The trainer LCD box (see Section 4.4) and the front panel **SPL** indicator show the sound pressure level for the next trial. If the next trial is a catch trial, the front panel **catch trial** indicator will light up and the **SPL** indicator will show zero. The trainer LCD will display "Catch trial".
- To manually force a specific SPL on the next trial, press the **Override SPL** button and type the desired SPL in the **Level (dB)** box before pressing the response button. The **Override SPL** button will remain depressed—if only a single trial is desired, the user must un-press the **Override SPL** button before pressing a response button.
- To force the next trial to be a catch trial, press the **Force catch** button before pressing the response button.
- If the subject responded before the tone, but the response is visible on the screen, the correct procedure is to press **early** (for the early response) and **no**, since there was no response within the window after the red bar. The only way there could be an early response and a yes would be if the subject responded twice—once just before the tone and again after the tone.

4.4 DIVE SCORING SYSTEM

Each dive is given a numerical score based on the performance of the subject. The purpose of the score is to provide the trainer with a metric that may be used to gauge the amount of reinforcement per dive. The score does not tell the trainer exactly what kind and how much reinforcement to give, rather the score simply gives the trainer a relative feel for whether the dive was "good" or "bad." It is up to the trainer to consistently apply the score from dive to dive and session to session to ensure that the subject's reinforcement is consistent with a given level of performance.

The subject's performance during each dive is quantified by counting the number of hits, misses, FAs, CRs, and any responses that may have been outside a trial window. The individual points awarded for each hit, miss, CRs, and FAs are specified in the Session Configuration dialog (see Section 5.3). The total score for the dive is the sum of points from each hit, miss, etc. Responses outside of a trial window are scored as FAs.

4.5 RESPONSE DETECTOR

The HTP includes an algorithm to automatically detect the onset of a subject's response. This algorithm allows one to calculate the response latency, defined as the time between the stimulus onset and the response onset. In mammals, the response latency is correlated with the perceived loudness of a sound, thus measurements of response latency allow estimates of the loudness of sounds.

The response detector is based on an energy detection technique. Figure 17 illustrates the steps involved in the response detection. The digitized pressure waveform (Figure 18(A), Figure 19(B)) is first band-pass filtered, then squared. Next, the squared waveform (Figure 18(B)) is broken into smaller time sequences, whose lengths are determined by the **time resolution** parameter. The squared pressure within each time sequence is then summed. The sum of the squared pressure is analogous to the energy in the sound, so this last step is equivalent to summing the energy in each small time sequence. This energy summation is repeated for each time sequence and the results used to generate a plot of the sound energy as a function of time. The logarithm of these data is then calculated. The logarithm operation has the effect of exaggerating the separation between the data at the lower range of values—this is important since we are trying to determine exactly when the response rises above the background noise. The log data (Figure 19(B)) are then compared to the user-defined **threshold** value and **minimum time** parameters. The algorithm searches for the first time value at which the log energy is above the specified threshold and stays above that threshold for a time duration equal to or greater than the minimum time width. This last feature is used to excludes transient response such as echolocation clicks.

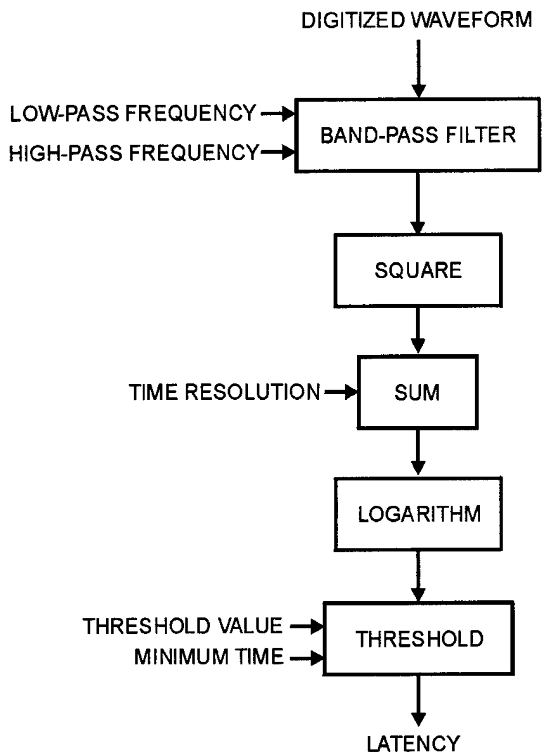


Figure 17. Response detector block diagram.

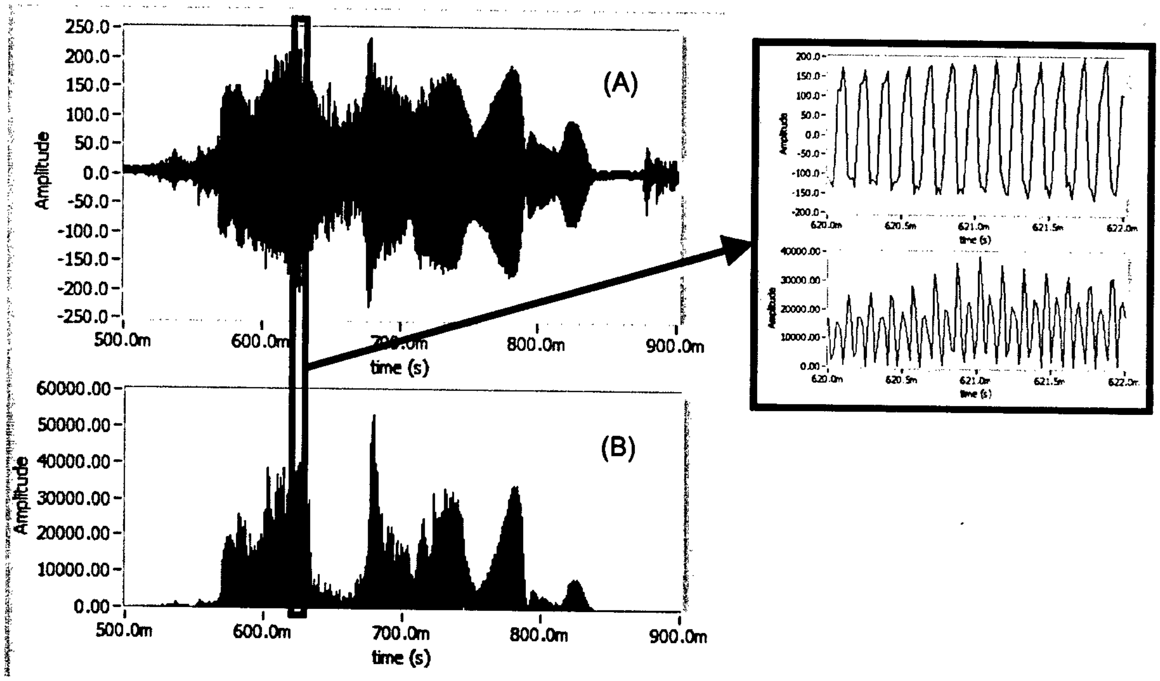


Figure 18. Response detector squaring operation. (A) Response instantaneous pressure waveform. (B) Instantaneous pressure waveform squared.

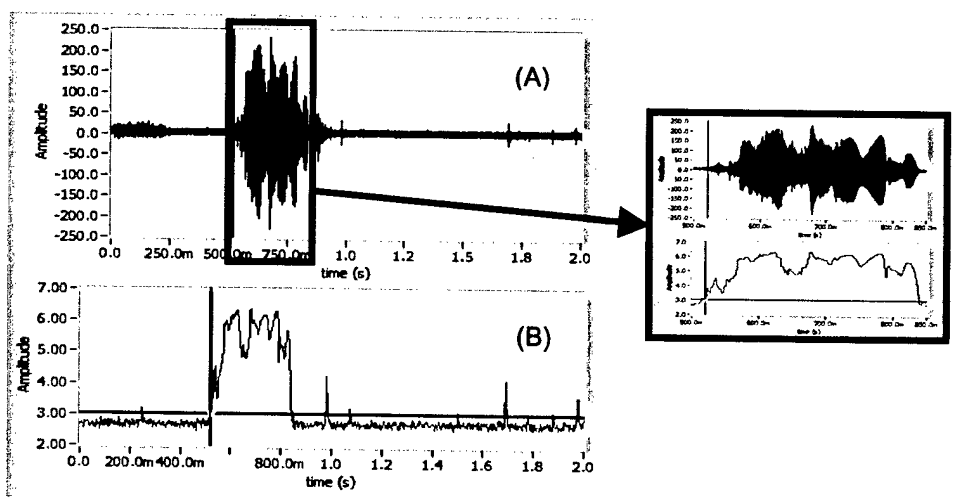


Figure 19. Response detector integration operation. (A) Response instantaneous pressure waveform. (B) Sum of squared instantaneous waveform displayed on log scale.

4.6 TRAINER LCD

A liquid crystal display (LCD) provides feedback to the trainer regarding the dive and trial parameters. The LCD is updated only at specific times. It indicates the update times (events) and what the LCD displays. Table 3 provides a summary of the Trainer LCD display.

Table 3. Trainer LCD display summary.

Event	Display
*HTP.EXE start	HTP
Session start	Ready ...
Operator presses control box S1 send button	Sending S1 S1 Sent
S2 (catch) trial start	trial/cumTrial mm:ss Catch VR ***vr***
S2 (tone) trial start	trial/cumTrial mm:ss ***SPL*** dB VR ***vr***
S2 trial end	trial/cumTrial mm:ss End VR ***vr***
Dive end	trial/cumTrial mm:ss Score **score**
Session end	Closing session ...

trial is the trial number within the current dive

cumTrial is the cumulative trial number

mm:ss is the dive time in minutes:seconds

SPL is the S2 tone SPL

vr is the number of correct responses required for the current dive (for the VR schedule)

score is the score from the dive

5. HTP.EXE REFERENCE

5.1 FRONT PANEL REFERENCE

The HTP front panel is divided into three main areas (Figure 20). The left side contains a number of controls and indicators regarding the status of the session, the current dive, and the next dive. The top area contains a plot of the session staircase showing the previous trials and whether they were hits, misses, false alarms, or correct rejections. The bottom region contains a plot of the last trial, which shows the subject's whistle response, if present. Table 4 lists the specific controls and indicators on the HTP front panel.

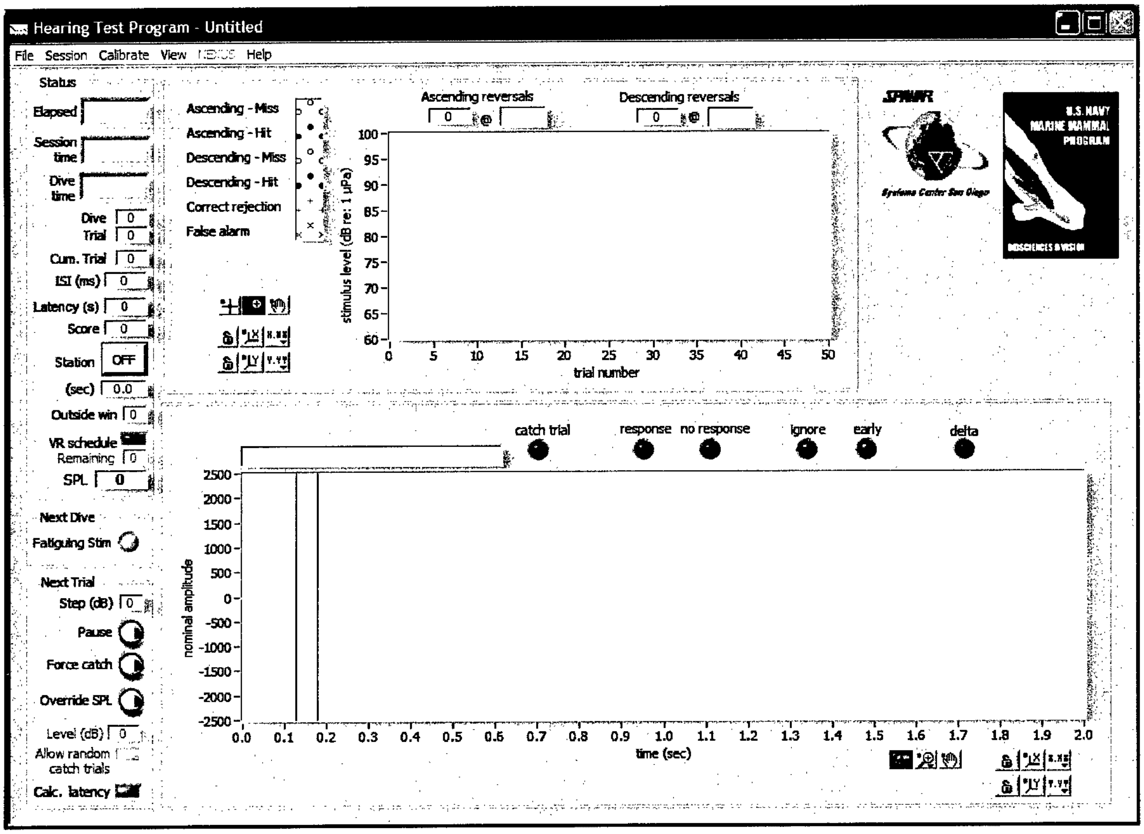


Figure 20. HTP front panel.

Table 4. HTP front panel controls and indicators.

Control/Indicator	Description
Elapsed	Elapsed time from the last session started with the **Clear elapsed time** box checked.
Session time	Elapsed time since the start of the current session.
Dive time	Elapsed time since the **Station** button was pressed ON.
Dive	Current dive (starts with zero).

Table 4. HTP front panel controls and indicators. (continued)

Control/Indicator	Description
Trial	Current trial (starts with zero).
Cum. Trial	Total number of trials (starts with zero).
ISI	Inter-Stimulus Interval–the randomized amount of time before the next trial.
Latency	Estimated time between the tone onset and the start of the subject's response. This indicator is only updated if the **Calc. latency** button is lit.
Score	Tabulated score from the previous dive (updated only after the dive ends).
Station	ON/OFF button showing if the subject is on or off the S2 station. This button is a control and an indicator. The function key F1 is mapped to this button. This indicator is also mapped to the Control Box **On S2** button.
(sec)	Cumulative time that the subject has spent on the S2 station.
Outside win	Cumulative responses outside the response window (i.e., before or after an actual trial). This indicator is mapped to the Control Box **Early** button.
VR schedule	Indicates if the variable ratio (VR) schedule is in effect. May be used as a control and an indicator.
Remaining	If VR schedule is in effect, indicates the number of correct responses required in the current dive before reinforcement.
SPL	SPL of the upcoming trial. If the current trial is a catch trial, **SPL** will display 0.0.
Fatiguing Stim	If this button is depressed, then one (or more) FS tones will be generated when the S1 trigger is pressed. When pressed, this button turns red. For this button to be accessible, **Use FS** must have been checked during session configuration.
Step	Amount of dBs that the S2 tone SPL will be decreased after each hit or increased after each miss. This object is a control and an indicator.
Pause	Depress this button to pause the program during a dive. Depress again to resume testing. This button is used infrequently. One example would be if the subject's stationing posture deteriorated during a dive and you wished to pause S2 tone presentations until stationing improved. *This button must be depressed before the response of the previous trial is keyed in, otherwise the trial will start.* The function key F12 is mapped to this button.
Force Catch	Depress this button to force the next trial to be a catch trial. At the conclusion of the trial, the button will pop out to the default state; that is, only one catch trial will be generated. To generate successive catch trials, the button must be depressed before each trial. *Note that this button must be depressed before the response of the previous trial is keyed in.* The function key F5 is mapped to this button.
Override SPL	Depress this button to force the next tone trial to be at a specific SPL level. The function key F7 is mapped to this button.
Level	This control is used to define the specific SPL value to generate when the **Override SPL** is depressed.
Allow random catch trials	If this button is lit, then random catch trials will be allowed to occur (if the Catch trials percentage in the session configuration window is greater than zero). If this button is not lit, then no catch trials will occur—all trials will be at the specified forced level. This control is only accessible if **Override SPL** is depressed. This button acts as a control and an indicator.

Table 4. HTP front panel controls and indicators. (continued)

Control/Indicator	Description
Calc. latency	Depress this button to automatically calculate the response latency after each trial. The result is displayed in the **Latency** indicator and with the green line in the response plot.
Ascending reversals	Current number of ascending reversals and the mean time at which the reversals occurred (if less than 10 reversals) or the mean time at which the last ten reversals occurred.
Descending reversals	Current number of descending reversals and the mean time at which the reversals occurred (if less than 10 reversals) or the mean time at which the last ten reversals occurred.
catch trial	Lights green if the upcoming trial is a catch trial.
response	Lights green when the operator presses the Control Box **YES** button after a trial.
no response	Lights green when the operator presses the Control Box **NO** button after a trial.
ignore	Lights yellow when the operator presses the Control Box **IGNORE** button after a trial.
early	Lights yellow when the operator presses the Control Box **EARLY** button anytime except during the actual tone representation and response recording (the operator is "locked-out" during this time period). Pressing the Control Box **EARLY** button increments a counter used to keep track of the total number of false alarms and "out-of window" responses. The false alarm rates are calculated based on the number of these responses that occur only during the trials containing the desired reversals, therefore it is important to key in the early responses close to the trial where they actually occurred.
delta	Depress this button to indicate that a "delta" was given to the subject or the subject was recalled during or at the conclusion of the trial. This button must be depressed before the response button. Trials marked with as having a delta are recorded in the session SDF and displayed differently in the session printouts.

5.2 MENU REFERENCE

Tables 5 through 10 describe the six HTP front panel menus: **File**, **Session**, **Calibrate**, **View**, **NEXUS**, and **Help**.

Table 5. HTP front panel **File** menu options.

Option	Description
Open Datalog	Opens a separate window within which the user can view, import, export, or edit the contents of the datalog file. See Section 8 for more information about the datalog file.
Get Directory Size	Calculates the size of the Base Path folder (and all subfolders) and displays the results in a separate window.
Clean Directories	Opens a separate window within which the user can delete specific files from within the Base Path folder and subfolders. Use Shift-click to select multiple directories and file types. Press the Filter button to generate a list of all files within the selected folders with the selected extensions. Pressing the Delete button will permanently delete all files appearing in the list. The Add to list buttons may be used to add custom folders and file extensions to the lists (if necessary).
Open Calculator	Launches the Microsoft® Windows calculator application.

Table 6. HTP front panel **Session** menu options.

Option	Description
Begin New Session	Opens the Session Configuration dialog and starts a test session.
End Session	Closes the current test session.
Edit Session	
Edit Comments	Opens a separate window within which the user can add/modify any session comments originally entered before or after a test session.
Rename	Opens a separate window within which the user can rename an existing session. This option may be used if an animal code is entered incorrectly.
Analyze Session Data	Open an existing SDF file to calculate hearing thresholds. See Section 6 for more information.
Compare Multiple Sessions	Plot results from multiple sessions on a single graph of hearing threshold as a function of the time relative to some reference time. For TTS testing, the reference time is the time of exposure to the fatiguing stimulus. See Section 7 for more details.

Table 7. HTP front panel **Calibrate** menu options.

Option	Description
S1 and FS	
Tonal	Opens the S1 and FS Tone Calibration window (Section 9).
Impulsive	Opens the S1 and FS Impulse Calibration window (Section 12).
S2	Opens the S2 Calibration window (Section 10).
Noise	Opens the Noise Calibration window (Section 11).

Table 8. HTP front panel **View** menu options.

Option	Description
Preferences	Opens the Session Configuration window without starting a test session. This option is useful to set up the required test parameters before a session.
EVT file	Opens an EVT file for viewing/editing with WordPad.
SDF file	Opens an SDF file for viewing/editing with WordPad.
SRC file	Opens an SRC file for viewing/editing with WordPad.
S1 binary	Opens an S1 recording and displays the waveforms in the front panel response window.
S2 binary	Opens an S2 recording and displays the waveform in the front panel response window.
Staircase data	Opens an existing SDF and displays the staircase on the front panel.
Date codes	Opens a separate application showing the conversion from dates to three character date codes.
Control Panel	Opens a small window containing a "virtual" Control Box (Figure 21). The virtual Control Box functions exactly as the physical device. This option is primarily used for troubleshooting/debugging on a computer without a hardware control box.

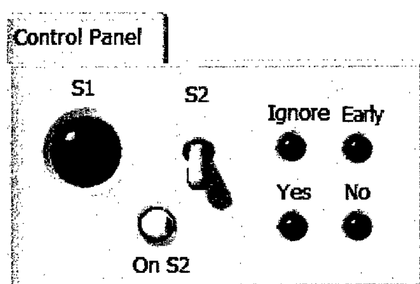


Figure 21. HTP "virtual" Control Box.

Table 9. HTP front panel **NEXUS** menu options.

Option	Description
Configuration	Opens the NEXUS configuration window (Figure 22), which allows the user to save/load NEXUS setups, edit configuration parameters, view peak meters, and sleep/wake the NEXUS device. Only accessible if the NEXUS amplifier is connected to COM 1 or COM 2.
Sleep	Puts the NEXUS into a low-power sleep mode. Requires the NEXUS to be connected to COM 1 or COM 2.
Wake	Returns the NEXUS from low-power "sleep" mode to normal status. Requires the NEXUS to be connected to COM 1 or COM 2.

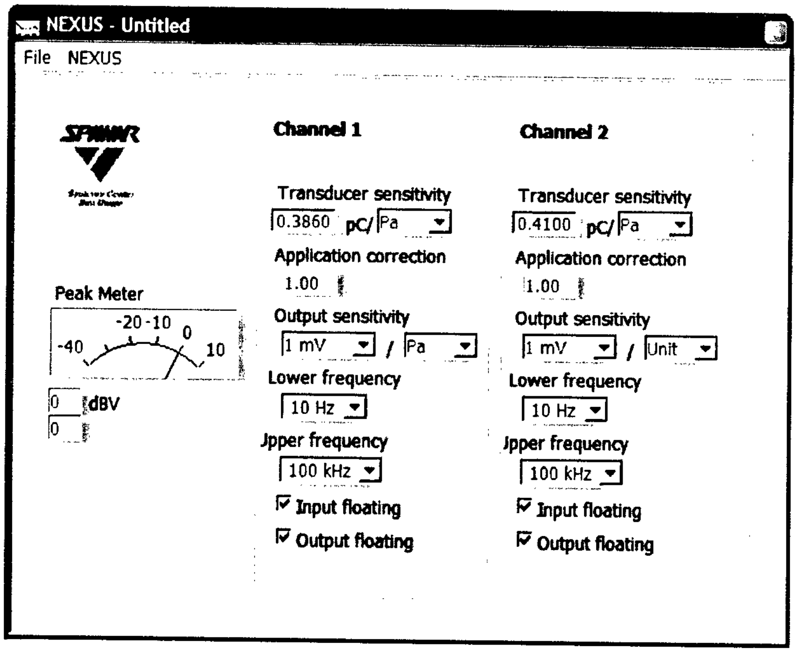


Figure 22. NEXUS configuration window.

Table 10. HTP front panel **Help** menu options.

Option	Description
Online Reference	Opens a PDF version of this document.
About HTP	Shows the HTP.EXE version number and author contact information (Figure 23).

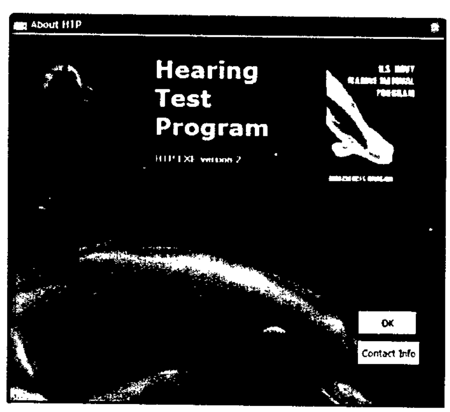


Figure 23. HTP information displayed after selecting **About HTP**.

5.3 SESSION CONFIGURATION DIALOG

The Session Configuration dialog appears when the operator selects **Begin New Session** from the **Session** menu or **Preferences** from the **View** menu. In the former case, after pressing OK, a session will begin using the parameters shown; in the latter case, a session will not be started, but the parameters will be retained. Using the **View: Preferences** menu item is a convenient way to set up the session parameters ahead of time.

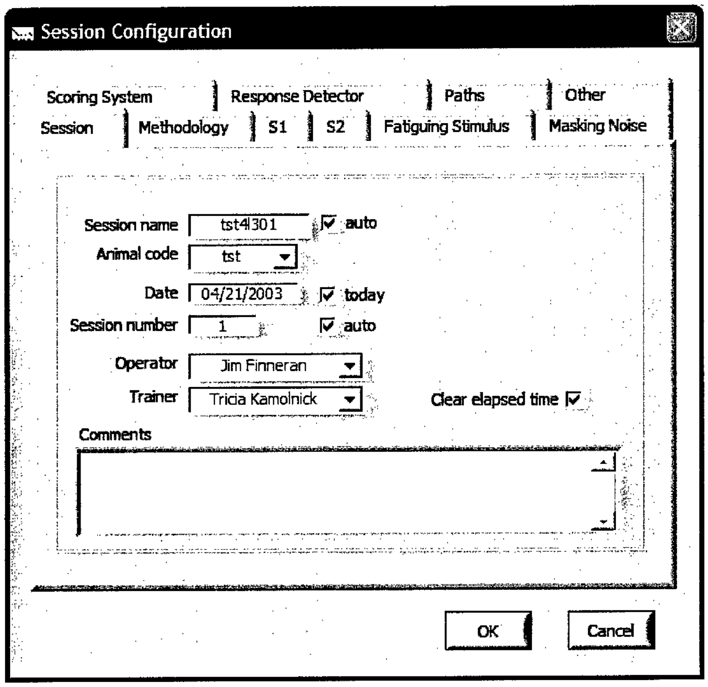


Figure 24. Session configuration dialog.

The Session Configuration dialog contains 10 tabs: **Session, Methodology, S1, S2, Fatiguing Stimulus, Masking Noise, Scoring System, Response Detector, Paths**, and **Other**. The options available on each tab are described in Tables 10 through 19.

Table 11. Session configuration **Session** tab.

Option	Description
Session name	Name of this session. This will be used as the file prefix for the SDF, EVT, S1, and S2 data files. If the **auto** box is checked, **Session name** will be automatically generated from the **Animal code**, **Date**, and **Session number**. The date is converted to a three character code of the form "mdy", where m = month, d = day, and y = year. Days and months greater than 9 are coded using letters, e.g., 10 = a, 11 = b, etc. For the year, 1 = 2001, 2 = 2002, etc.
Animal code	Three letter code used to identify the subject. The code "tst" indicates a test session, i.e., no subject present. Test session files may be easily deleted when the HTP program is exited. Available animal codes are stored in a text file named subjects.txt in the TEXT folder.

Table 11. Session configuration **Session** tab. (continued)

Option	Description
Date	If the **today** box is checked, the current date will be automatically entered.
Session number	Number of current session. If the **auto** box is checked, the next available number will be used.
Operator	Name of computer operator. Available names are stored in a text file named operators.txt in the TEXT folder.
Trainer	Name of trainer. Available names are stored in a text file named trainers.txt in the TEXT folder.
Clear elapsed time	If this box is checked, the **Elapsed** time indicator on the front panel will be reset to zero at the start of the session. Normal operation is to check this box for the first session run with a subject. The box is then cleared for subsequent sessions so that the total elapsed time is shown (i.e., the total time since the start of the first session).
Comments	Text comments about the session.

Table 12. Session configuration **Methodology** tab.

Option	Description
Staircase type	Used to select the type of staircase procedure: ascending, descending, ascending and descending (alternating), or ascending and descending (random).
Catch trials	Percentage of trials with no S2 tone (catch trials).
Initial step size, Final step size	Amount (in dB) to decrease the S2 tone amplitude after a hit and to increase after a miss. The **Initial step size** is used from the session start until the first reversal. The **Final step size** is used after the first reversal.
Minimum ISI, Maximum ISI	Minimum and Maximum interstimulus interval (ISI), the time between S2 tones.
Warm-up trials?	If this box is checked, a number of warm-up trials will be presented at the beginning of the session at a fixed SPL.
Number	Number of warm-up trials.
SPL	SPL of warm-up trials.
VR schedule	If this box is checked, the PC will count the number of correct responses and display the number of correct responses remaining until reinforcement. The specific number of correct responses required for a dive is chosen randomly from the list of numbers in the **Ratios** control.
Ratios	The numbers to use for the VR schedule. Numbers should be separated by commas.

Table 13. Session configuration **S1** tab.

Option	Description
S1 out device	Device number used to generate S1 signals.
S1 DAC	Channel number for S1 signals.
S1 max update rate	Rate at which digital values are written to the output channel. This number should be at least 5 to 10 times the S1 signal frequency. The same value must be used for calibration and the session.
S1 source file	File path for the S1V file used to specify the S1 signal frequency, amplitude, duration, rise time, delay, AM frequency, and AM amplitude.
View S1 source file	Allows the operator to view an S1V source file (these are text files).
S1 in device	Device number for the S1 hydrophone recording.
Min. delay	Minimum value for random interval from button press to S1 generation.
Max. delay	Maximum value for random interval from button press to S1 generation.
S1 ACH	Analog input channels for the S1 hydrophones mounted on the biteplate or frame. Channel numbers are separated with a comma.
S1 AI duration	Duration of sound recording during S1 signal presentation.
S1 AI scan rate	Scan rate during S1 signal presentation.
S1 sensitivity	Sensitivity of S1 hydrophone/charge amp in mV/Pa.
S1 AI limits	Maximum expected voltage range for S1 hydrophone/charge amp output.

Table 14. Session configuration **S2** tab.

Option	Description
S2 out device	Device number used to generate S2 tones.
S2 DAC	Channel number for S2 tones.
S2 max update rate	Rate at which digital values are written to the output channel. This number should be 5 to 10 times the S2 tone frequency. The same value must be used for calibration and the session.
S2 source file	File path for the SRC file used to specify the S2 tone frequency and amplitude.
View S2 source file	Allows the operator to view an SRC file (these are text files).
Starting SPL (ascend)	Starting SPL for ascending staircase. The available values are based on those found in the selected SRC file.
Starting SPL (descend)	Starting SPL for descending staircase. The available values are based on the those found in the selected SRC file.
Delay	Time delay from trial recording start to S2 tone start.
Duration	Duration of S2 tones.

Table 14. Session configuration **S2** tab. (continued)

Option	Description
Rise/fall	Rise and fall time for S2 tones. The tone rise and fall are linear.
S2 in device	Device number for the S2 hydrophone recording.
S2 ACH	Analog input channel for the S2 hydrophone. If more than one channel is specified (using numbers separated by commas), the first channel listed will be used.
S2 AI duration	Duration of each trial.
S2 AI scan rate	Scan rate during each trial recording.
S2 sensitivity	Sensitivity of S2 hydrophone/charge amp in mV/Pa.
S2 AI limits	Maximum expected voltage range for S2 hydrophone/charge amp output.

Table 15. Session configuration **Fatiguing Stimulus** tab.

Option	Description
Use FS	This box must be checked for a FS to be generated during a session.
FS out device	Device number used to generate FS.
FS DAC	Output channel number for FS.
FS max update rate	Rate at which digital values are written to the output channel. This number should be at least 5 times the FS frequency. The same value must be used for calibration and the session.
FS source file	File path for the FS waveform file. This file is created during the FS calibration process.
Number of FS tones	Number of FS tones to send with each S1 signal.
Time interval between FS	Time interval between each successive FS sent (for multiple FS).
FS recording duration	Time duration for hydrophone recordings during the FS presentation.

Table 16. Session configuration **Masking Noise** tab.

Option	Description
Use masking noise	This box must be checked for masking noise to be generated during a session.
Noise out device	Device number used to generate masking noise.
Noise DAC	Channel number for masking noise.
Noise source file	File path for the noise FIR file used to specify the masking noise amplitude, update rate, and digital filter.
Noise duration	Duration of masking noise.
Noise rise/fall	Rise and fall time for masking noise.

Table 17. Session configuration **Scoring System** tab.

Option	Description
Show trainer score	Determines whether the score from each dive is displayed on the trainer LCD. The score from a particular dive is the sum of the points from each trial. Each trial's point value is determined by the outcome of the trial and the numerical values entered in **Hit pints**, **Miss points**, **CR points**, and **FA points**. For example, if the trial results in a HIT, the trial score is equal to the value in the **Hit points** control.
Hit points	Number of points to assign to each HIT (response to a tone).
Miss points	Number of points to assign to each MISS (no response to a tone).
CR points	Number of points to assign to each CORRECT REJECTION (no response to a catch trial).
FA points	Number of points to assign to each FALSE ALARM (response to a catch trial) and each response that occurs outside of a trial window.
Multiplier	Final score from each dive is multiplied by **Multiplier** before being displayed. Use this if a simple conversion is found from the raw score to the number of fish or "fish units".

Table 18. Session configuration **Response Detector** tab.

Option	Description
Show response latency	Determines if the response latency is automatically calculated and displayed on the HTP front panel during a session. This parameter may be toggled during a session using the front panel **Calc. latency** control.
HP Filter	Response detector high-pass filter setting. Must be lower than the fundamental frequency in the response. The default setting for dolphins is around 4 kHz.
LP Filter	Response detector low-pass filter setting. Must be higher than the fundamental frequency in the response. For dolphins, a typical setting is 20 kHz.
Threshold	Response detector threshold (dimensionless). Default value is 3.0.
Min. width	Minimum time that energy level must exceed **Threshold** for a response to be considered valid. Values are typically in the 10-20 ms range.
Latency resolution	Time sequence duration for energy calculations. The squared-pressure waveform is quantized into time sequences having this length. Large values for the resolution will produce very clear response onsets, but lower the precision of the latency estimate. Note that the final reported latency measures are constrained by the **Latency resolution**.
Response button default	Default state of control box response buttons. Some push-button switches are OPEN by default, others are closed; either will work on the control box, but they must all be the same type and the default state must be selected here.

Table 19. Session configuration **Paths** tab.

Option	Description
Base path	Location of the main HTP folder.
Datalog filename	Filename for the Datalog. This string should contain the actual filename only, not the folder or path structure.

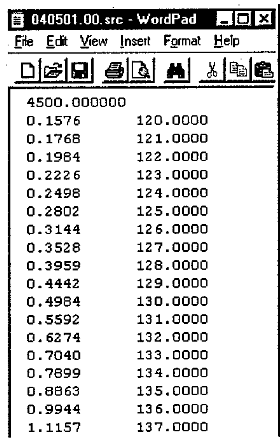
Table 20. Session configuration **Other** tab.

Option	Description
LCD	Serial port for trainer LCD box.
NEXUS	Serial port for NEXUS charge amp.
Location	Text string specifying the location (e.g., Pools, Tacoma, etc.).
Catch trial display level	SPL at which the catch trials will be displayed on the front panel staircase plot.

5.4 DATA FILE FORMATS

5.4.1 SRC Files

SRC ("source") files are created during the S2 calibration process. The computer uses these files to translate a desired SPL into a specific voltage produced at the DAC. SRC files are ASCII tab-delimited text files. The first row indicates the frequency. The remaining rows consist of pairs of numbers separated by a TAB. The first number indicates the voltage, the second number gives the SPL produced from that voltage. Figure 25 shows an example of an SRC file.

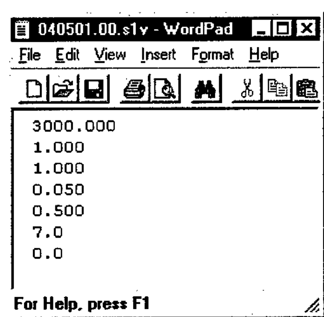



```
4500.000000
0.1576	120.0000
0.1768	121.0000
0.1984	122.0000
0.2226	123.0000
0.2498	124.0000
0.2802	125.0000
0.3144	126.0000
0.3528	127.0000
0.3959	128.0000
0.4442	129.0000
0.4984	130.0000
0.5592	131.0000
0.6274	132.0000
0.7040	133.0000
0.7899	134.0000
0.8863	135.0000
0.9944	136.0000
1.1157	137.0000
```

Figure 25. Example SRC file.

5.4.2 S1V Files

S1V ("S1 voltage") files are created during the S1 calibration process. These files store the information necessary to create the S1 signal. S1V files are ASCII tab-delimited text files consisting of a single column of seven numbers indicating the S1 frequency, amplitude (i.e., voltage), duration (in seconds), rise and fall times (in seconds), minimum delay (in seconds) from S1 send button to actual sound, modulation (AM) frequency (in Hz), and modulation amplitude (%). Figure 26 shows an example S1V file.




```
3000.000
1.000
1.000
0.050
0.500
7.0
0.0
```

Figure 26. Example S1V file.

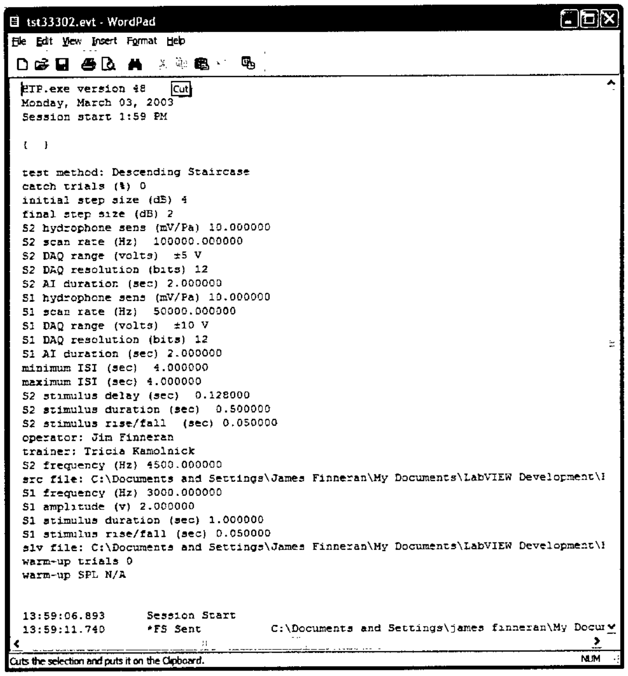
5.4.3 SDF Files

SDF files are created during the test sessions and contain a record of every trial. SDF files are ASCII tab-delimited text files with 16 columns and a variable number of rows. Rows 1 through 16 contain text descriptions of the data stored in each column (also called a "field"):

FIELD00 TIME (HH:MM:SS.SSS)
FIELD01 DIVE NUMBER
FIELD02 TONE NUMBER (WITHIN DIVE)
FIELD03 TONE NUMBER (CUMULATIVE)
FIELD04 FREQUENCY (HZ)
FIELD05 S2 AMP (VOLTS)
FIELD06 NOMINAL STIMULUS LEVEL
FIELD07 ACTUAL RESPONSE (0=NO,1=YES,2=IGNORE,3=EARLY)
FIELD08 STIMULUS PRESENT (0=NO,1=YES)
FIELD09 STAIRCASE TYPE (0=UP,1=DN)
FIELD10 SOUND PRESSURE LEVEL (DB RE 1 μPA)
FIELD11 DELTA (0=NO,1=YES)
FIELD12 OUT-OF-WINDOW RESPONSES (CUMULATIVE)
FIELD13 TIME ON S2 STATION (CUMULATIVE)
FIELD14 UNTITLED
FIELD15 UNTITLED

5.4.4 EVT Files

EVT ("event") files are also created during the test sessions. EVT files are ASCII text files that contain a record of certain session parameters, the time that each S1 and S2 signal is generated, and any comments entered by the operator. Figure 27 shows an example EVT file.



tst33302.evt - WordPad

File Edit View Insert Format Help

```
HTP.exe version 48
Monday, March 03, 2003
Session start 1:59 PM

{ }

test method: Descending Staircase
catch trials (%) 0
initial step size (dB) 4
final step size (dB) 2
S2 hydrophone sens (mV/Pa) 10.000000
S2 scan rate (Hz)  100000.000000
S2 DAQ range (volts)  ±5 V
S2 DAQ resolution (bits) 12
S2 AI duration (sec) 2.000000
S1 hydrophone sens (mV/Pa) 10.000000
S1 scan rate (Hz)  50000.000000
S1 DAQ range (volts)  ±10 V
S1 DAQ resolution (bits) 12
S1 AI duration (sec) 2.000000
minimum ISI (sec)  4.000000
maximum ISI (sec) 4.000000
S2 stimulus delay (sec)  0.128000
S2 stimulus duration (sec)  0.500000
S2 stimulus rise/fall  (sec) 0.050000
operator: Jim Finneran
trainer: Tricia Kamolnick
S2 frequency (Hz) 4500.000000
src file: C:\Documents and Settings\James Finneran\My Documents\LabVIEW Development\
S1 frequency (Hz) 3000.000000
S1 amplitude (v) 2.000000
S1 stimulus duration (sec) 1.000000
S1 stimulus rise/fall (sec) 0.050000
slv file: C:\Documents and Settings\James Finneran\My Documents\LabVIEW Development\
warm-up trials 0
warm-up SPL N/A


13:59:06.893      Session Start
13:59:11.740      *FS Sent        C:\Documents and Settings\james finneran\My Docu
```

Cuts the selection and puts it on the Clipboard. NUM

Figure 27. Example EVT file.

5.4.5 FIR Files

FIR files are created during the noise calibration process. The computer uses these files to generate masking noise with specific spectral parameters. FIR files are ASCII files that contain detailed information regarding the noise-frequency content, amplitude, duration, calibration results, and certain front panel settings.

5.4.6 FS Files

FS ("fatiguing stimulus") files store the FS waveform defined during the FS calibration process. FS files are single channel, signed, 16-bit binary files (Motorola format).

5.4.7 INI Files

The Microsoft® Windows operating system uses INI files to store configuration parameters for programs. The HTP.EXE program uses the configuration file HTP.INI, located in the EXE folder, to store default front panel control values and other setup parameters. INI files are ASCII text files and may be viewed using a text editor. Note, however, that manual changes to the HTP.INI file may result in unpredictable program behavior.

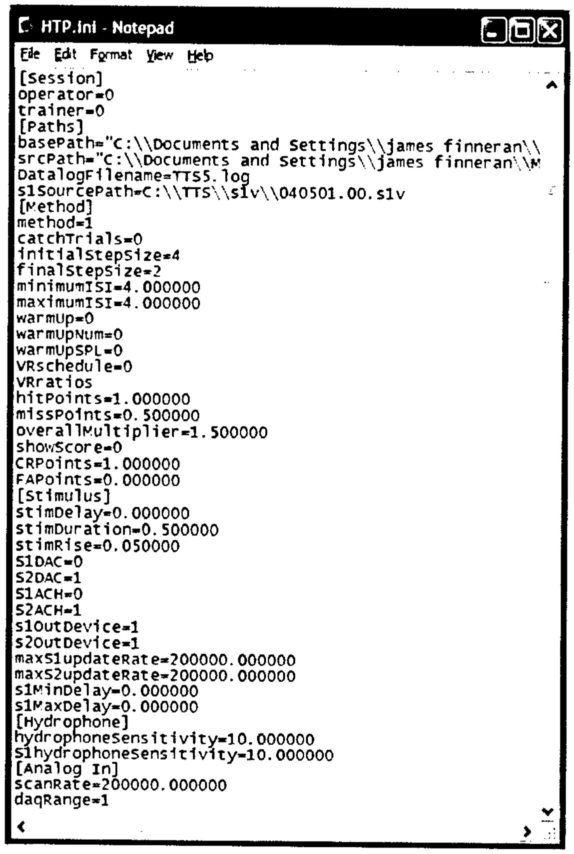

```
[Session]
operator=0
trainer=0
[Paths]
basePath="C:\\Documents and Settings\\james finneran\\
srcPath="C:\\Documents and Settings\\james finneran\\M
DatalogFilename=TTS5.log
s1SourcePath=C:\\TTS\\s1v\\040501.00.s1v
[Method]
method=1
catchTrials=0
initialStepSize=4
finalStepSize=2
minimumISI=4.000000
maximumISI=4.000000
warmUp=0
warmUpNum=0
warmUpSPL=0
VRschedule=0
VRratios
hitPoints=1.000000
missPoints=0.500000
overallMultiplier=1.500000
showScore=0
CRPoints=1.000000
FAPoints=0.000000
[Stimulus]
stimDelay=0.000000
stimDuration=0.500000
stimRise=0.050000
S1DAC=0
S2DAC=1
S1ACH=0
S2ACH=1
s1OutDevice=1
s2OutDevice=1
maxS1updateRate=200000.000000
maxS2updateRate=200000.000000
s1MinDelay=0.000000
s1MaxDelay=0.000000
[Hydrophone]
hydrophoneSensitivity=10.000000
S1hydrophoneSensitivity=10.000000
[Analog In]
scanRate=200000.000000
daqRange=1
```

Figure 28. Example HTP.INI file.

5.4.8 MAT Files

MAT files are double-precision binary MATLAB® format files created by the MATLAB® save command and readable by the MATLAB® load command. They can be created on one machine and later read by MATLAB® on another machine with a different floating-point format, retaining as much accuracy and range as the disparate formats allow. They can also be manipulated by other programs, external to MATLAB®. The binary formats used vary depending on the size and type of any arrays. MAT files include variable names and numeric values.

6. ANALYZING SESSION DATA

6.1 OVERVIEW

An SDF data file is generated during each hearing test session in which S2 tones are presented. This data file contains the test parameters (e.g., tone or catch trial, SPL level, etc.) and results (response or no response, response latency, etc.) for each individual trial. To actually calculate a hearing threshold, this "raw" data file must be analyzed using the **Session: Analyze Session Data** menu option from the main HTP.EXE front panel. After selecting this menu option, a dialog box will open asking the user to select the individual SDF file to analyze. The default file is the last session that was conducted. Select the desired file from the list. A new window will open showing the Analyze Session front panel (Figure 29). The appropriate calculations will be performed and the results automatically displayed.

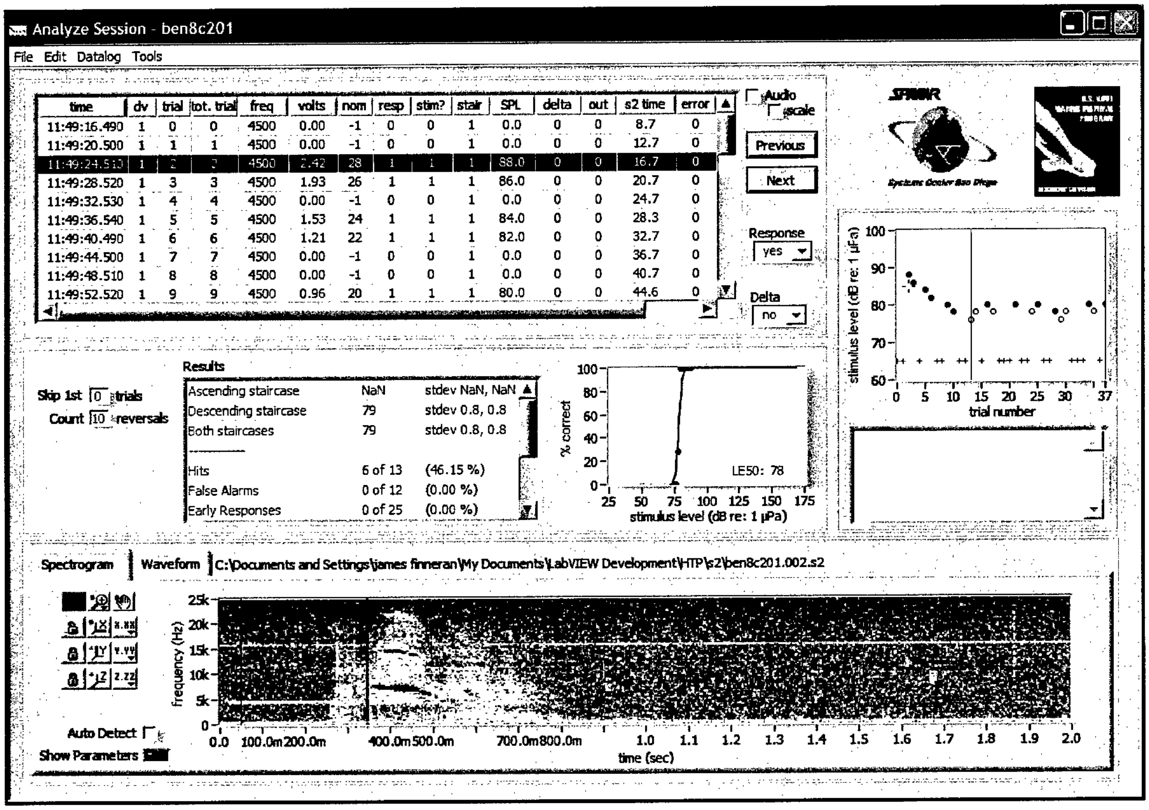
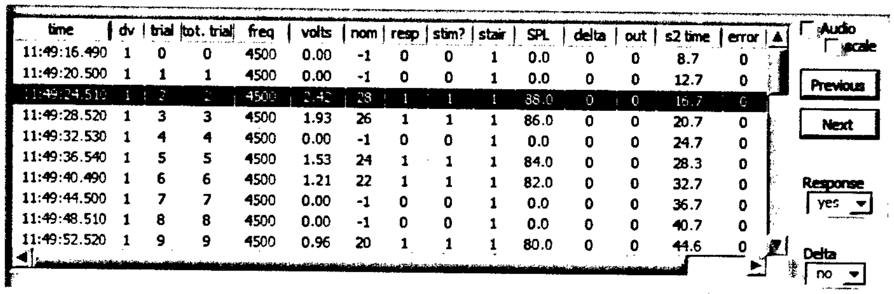


Figure 29. Analyze Session front panel.

Auditory thresholds obtained using the staircase procedure are based on the concept of a "reversal", defined as any transition from a hit to a miss, or a miss to a hit. These points are called reversals because this is where the S2 SPL values change from decreasing to increasing or vice versa. Thresholds are defined as the mean SPL over some number of reversals, usually 10. Auditory thresholds have units of pressure; if the step size over the reversals is equal (i.e., same increase after a miss as a decrease after a hit) the threshold gives the SPL at which the subject would respond to the sound 50% of the time. It is important to remember that the hearing threshold is a statistical concept—just because a sound is at a received level above threshold does not guarantee that the subject will hear it every time it is presented.

6.2 FRONT PANEL REFERENCE

The upper left portion of the Analyze Session front panel contains a table showing the data from the SDF file (Figure 30). Each row of the table indicates the parameters associated with a specific trial. The blue highlighting indicates the currently selected trial; clicking in another row of the table will select a new trial. The scrollbars may be used to scroll through the table. The **Previous** and **Next** buttons to the right of the table advance the current trial backward or forward, respectively. The **Response** and **Delta** pull-down menus may be used to change the recorded response of a trial or whether a delta was given on that trial. If any changes are made, the file must be saved using the **File: Save** menu option. If a check is placed in the **Audio** checkbox, each new trial selected will be played back through the PC speakers. This option is useful to check if a response actually occurred on a specific trial. To scale all playbacks to maximum volume, check the **scale** box.



time	dv	trial	tot. trial	freq	volts	nom	resp	stim?	stair	SPL	delta	out	s2 time	error
11:49:16.490	1	0	0	4500	0.00	-1	0	0	1	0.0	0	0	8.7	0
11:49:20.500	1	1	1	4500	0.00	-1	0	0	1	0.0	0	0	12.7	0
11:49:24.510	1	2	2	4500	2.42	28	1	1	1	88.0	0	0	16.7	0
11:49:28.520	1	3	3	4500	1.93	26	1	1	1	86.0	0	0	20.7	0
11:49:32.530	1	4	4	4500	0.00	-1	0	0	1	0.0	0	0	24.7	0
11:49:36.540	1	5	5	4500	1.53	24	1	1	1	84.0	0	0	28.3	0
11:49:40.490	1	6	6	4500	1.21	22	1	1	1	82.0	0	0	32.7	0
11:49:44.500	1	7	7	4500	0.00	-1	0	0	1	0.0	0	0	36.7	0
11:49:48.510	1	8	8	4500	0.00	-1	0	0	1	0.0	0	0	40.7	0
11:49:52.520	1	9	9	4500	0.96	20	1	1	1	80.0	0	0	44.6	0

Figure 30. Detail of tabular data from the SDF as displayed in the Analyze Session front panel.

The right side of the Analyze Session front panel shows a plot of the staircase data as seen on the main front panel during a session. The data point corresponding to the currently selected trial has a cross superimposed on it. Two vertical greens bars indicate the collection of reversals over which the threshold and FA rate were calculated.

Below the staircase plot a text box shows any comments entered by the user during the session. *Note that user modifications to the text in this window will not be saved.* To add or edit comments for an existing session, use the **Session: Edit Session: Edit Comments** menu item.

The center of the front panel shows the tabular results (Figure 31). These results include the threshold and false alarm rates. Thresholds are shown for the ascending staircase, descending staircase, and the average threshold from both staircases. If only a descending or ascending staircase method was used, the other threshold will show the result "NaN", which stands for "not a number," meaning the number is undefined. Each threshold result gives the mean (in dB re 1 μPa) and standard deviation (in dB). Two numbers are given for the standard deviation (lower, upper) in order to display the standard deviation properly in decibels, which is a logarithmic unit. The table also shows the hit rate (number of hits divided by the number of tone trials), false alarm rate (number of FAs divided by the number of catch trials), and the number of early responses. These three parameters are calculated from the set of trials indicated by the green lines in the staircase plot; *when calculating these parameters, only the set of trials containing the reversals are used.* The table also shows the total number of "out of window" responses and the rT1 index, which is proportional to the number of out-of-window responses plus the number of FAs, divided by the total amount of time that the subject was on the S2 biteplate without a tone trial being present. The rT1 index provides a measure of the total number of FAs normalized with respect to the amount of time that the subject had an opportunity to commit a FA, i.e., the amount of time on the S2 station outside of a tone trial.

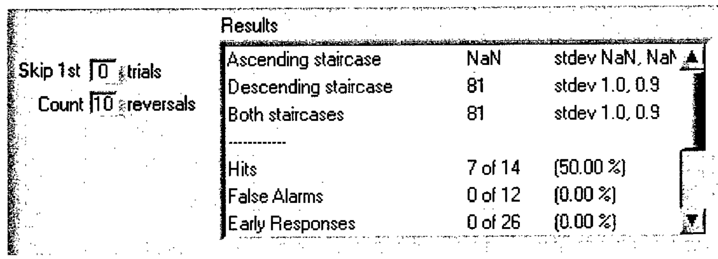


Figure 31. Analyze Session tabular results detail.

To the left of the table are two numerical controls that may be used to determine which trials are used in threshold calculations. Use the **Skip 1st** setting to skip a number of trials at the start of the session; this setting is normally used to exclude one or more high-level reversals followed by a much lower plateau in the staircase. The **Count** control is used to set the number of reversals used in the calculations. Some trial and error may be necessary to find the exact number of reversals in sessions with more than 10 reversals. If a session contains less than 10 reversals, the **Count** must be changed to the exact number of reversals.

To the right of the table is a plot of the percent correct detected versus the stimulus level (i.e., the S2 tone SPL). These data are obtained by pooling all tone trials with the same SPL and then calculating the percentage that were hits. The blue line is an ogive (S-shaped curve) fit to the data. The 50% correct detection point is the intersection of a horizontal line at the 50% correct detection point with the ogive fit to the data. If the data are well-behaved, the 50% correct point (LE50) should be approximately equal to the calculated threshold. The data will often not be well-behaved, because the staircase method tends to result in many trials at threshold but very few well above or below threshold, so low and high correct detection percentages may not exist or may be based on very few data points.

At the bottom of the front panel is a plot showing the hydrophone recording made during the trial (Figure 32). The display may be changed from the waveform display (sound pressure versus time, as seen during the session) to a spectrogram display (SPL as a function of frequency and time). The spectrogram also contains a black vertical line to indicate the response start time. The move cursor tool may be used to drag the response start indicator to different time values. Any changes made to the response start cursor are reflected in the SDF data table at the top of the screen. Note that the SDF must be saved by the user (using the **File: Save** or **File: Save As** menu items) to preserve changes made using the move cursor tool. At the bottom left of the Spectrogram tab are the **Auto Detect** and **Show Parameters** controls. If the **Auto Detect** button is checked, the response detection algorithm will automatically run on the current trial recording and overwrite any response start time entered by the user with the move cursor tool. Otherwise, the detector will only run if the SDF latency value for the current trial is zero or "Inf", which means infinity, or no response (infinity is the value recorded by the detector algorithm when no response is detected). This value means that the detector will run only if the user has not changed the response start time using the move cursor tool.

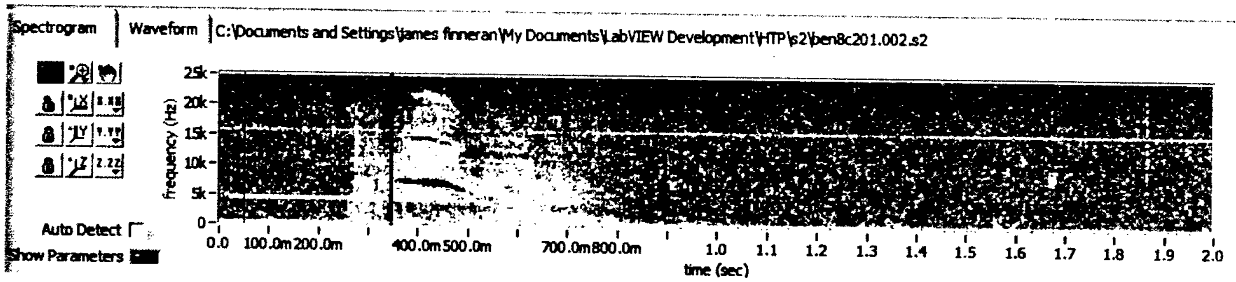


Figure 32. Hydrophone recording displayed in spectrograph format.

Pressing the **Show Parameters** button opens the Response Detector Parameters window (Figure 33). This panel allows the user to adjust the response detector parameters interactively. Changes made to the Detector Parameters values will cause the response detector to run on the current trial recording.

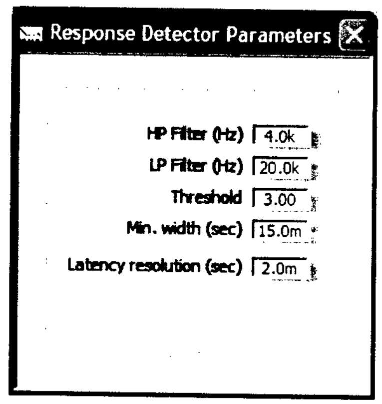


Figure 33. Response Detector Parameters window.

6.3 MENU REFERENCE

Tables 21 through 24 list the individual options for the Analyze Session **File**, **Edit**, **Datalog**, and **Tools** menus, respectively.

Table 21. Analyze Session **File** menu.

Option	Description
Open SDF	Produces a file open dialog box within which the user can select and open a new SDF file for analysis.
Save SDF	Save any changes made to the current SDF. Savable changes are (1) changes to a subject's response, (2) changes to the delta status, (3) SPL corrections, and (4) response latency measures. No other changes require saving the SDF. This option will replace the existing SDF with the data currently displayed in the table.

Table21. Analyze Session **File** menu. (continued)

Option	Description
Save SDF As	Save any changes made to the current SDF. Savable changes are (1) changes to a subject's response, (2) changes to the delta status, (3) SPL corrections, and (4) response latency measures. No other changes require saving the SDF. This option will create a new SDF with the data currently displayed in the table. Note that the new SDF file will be "disconnected" from other session files such as the EVT and S2 files, thus attempts to directly analyze the new SDF file may result in error messages.
Print	Print the session results.
Print Preview	Create a printable MATLAB® figure window containing the results. This option is useful for saving the printout as a graphic (TIF) file, or forcing the printer to print in pure black and white.
Close	Close the Analyze Session front panel and return to the main HTP.EXE front panel.

Table 22. Analyze Session **Edit** menu.

Option	Description
Correct SPL (dB) Values	Opens a dialog box within which the user may enter a number to use as a correction for the S2 tone SPLs. The correction is normally based on the error measured during postcalibration.

Table 23. Analyze Session **Datalog** menu.

Option	Description
View Datalog	Opens a separate window within which the user can view, import, export, or edit the contents of the datalog file. See Section 8 for more information about the datalog file.
Add Current Session	Add the results of the current session to the datalog. Every session that generates a threshold should be added to the datalog. If Show Add Dialog is checked, a dialog box will appear within which the user may view and/or modify the data that are being saved to the datalog. If the current session already exists in the datalog, the user will be prompted to replace the existing session or cancel the operation.
Show Add Dialog	If this menu item is checked, a dialog box will appear within which the user may view and/or modify the data that are being saved to the datalog before the save.
Prompt for Datalog Filename	If this box is checked, the user will be prompted for a datalog filename when adding the current session to the datalog. Otherwise, the session is added to the datalog file specified in the Preferences.

Table 24. Analyze Session **Tools** menu.

Option	Description
Preferences	Opens the Session Configuration window.

7. COMPARING MULTIPLE SESSIONS

7.1 OVERVIEW

It is often desirable to compare the results of two or more sessions. This comparison is accomplished using the front panel **Session: Compare Multiple Sessions** menu item. The Compare Multiple Sessions window (Figure 34) was created to allow the calculation and display of threshold shifts that may occur after exposure to intense sounds during TTS testing. In this application, the sessions being compared consist of the pre-exposure session and the post-exposure session. The difference between the post-exposure and pre-exposure thresholds is the amount of TTS. Although written specifically for TTS testing, this panel may be used to compare any two (or more) test sessions.

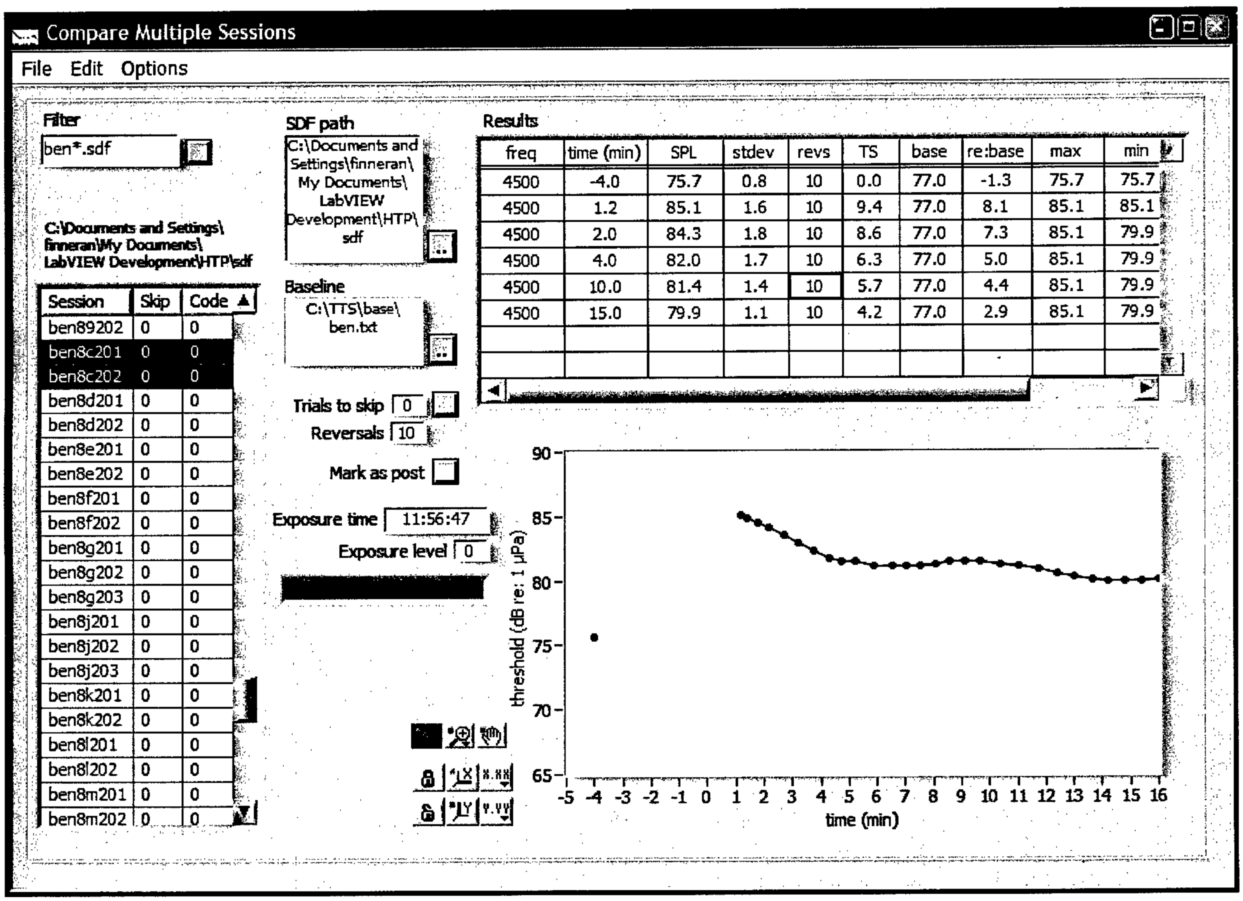


Figure 34. Compare Multiple Sessions front panel.

Selecting the **Session: Compare Multiple Sessions** menu option will open the Compare Multiple Sessions front panel. The **SDF path** control determines where the program looks for the SDF files. A list of all available SDF files will appear in a table on the left side of the screen. To filter this list, use the **Filter** text control. The "*" is used to specify any characters, thus, the **Filter** "ben*.sdf" will find all the files that begin with "ben" and end with ".sdf". Use the scrollbar to locate all of the SDF files for a particular subject and day. Left-click on a session name to select it. Shift-click to select multiple sessions.

Check the **Baseline** control to ensure that the correct baseline file for the current subject is listed; if not, use the browse button to the right of the control to select the proper file. The baseline file is an ASCII tab-delimited text file listing the subject's baseline hearing thresholds at various frequencies. The file requires two columns: the first contains the frequencies, the second contains the threshold at each frequency. The baseline thresholds are used to compare day-to-day results with a subject's long-term average hearing thresholds.

The number to the right of each filename in the table shows how many trials will be skipped when the data are analyzed. If any trials were skipped when the sessions were analyzed using the Analyze Session front panel, highlight the individual session name, type the number of trials that were skipped in the **Trials to skip** control, and press the button to the right of the control. The number to the right of the filename will change to reflect the new number of trials to skip. The **Reversals** control is used to set the number of reversals to use for threshold calculations (usually 10).

The **Mark as post** control is used to identify the post-exposure session. Left-click the post-exposure session in the table, then press the Mark as post button. This action will place a "1" in the third column ("Code") of the table, indicating which session is considered the post-exposure session.

Use the mouse and shift-click to select all of the SDF files for a particular subject and day. Use the **File: Analyze Selected Files** menu option (or **CTRL-O**) to analyze the files and display the tabular and graphical results. Finally, use the **File: Print** menu item to print the results.

When analyzing the selected sessions, the program uses the first S1 of the post-exposure session as the exposure time. The results show the threshold measured in each session and the time it was measured relative to the exposure time. The time for a threshold measurement is defined as the mean time of the reversals. The threshold SPL is the mean SPL over the reversals. For the post-exposure session, more than 10 reversals are normally obtained; for this session, a moving average is applied to the data to produce a curve of thresholds versus time. Thresholds at specific times post-exposure are determined by interpolating within the curve. The tabular data show the first and last thresholds measured and their times, as well as thresholds measured at 2, 4, and 10 minutes. The threshold shift (TS) is defined as the difference between the post-exposure and the pre-exposure thresholds. This number is the most important number for TTS testing.

7.2 FRONT PANEL REFERENCE

Table 25 describes the Compare Multiple Sessions front panel controls and indicators.

Table 25. Compare Multiple Sessions front panel controls and indicators.

Control/Indicator	Description
SDF path	Folder containing the SDF files. Use the browse button to the right of the control to change the default path.
Filter	This control is used to filter the list of all available SDF files into a smaller, more manageable list containing only sessions for a particular subject, for example. Use the '*' as a wildcard to match any text.

Table 25. Compare Multiple Sessions front panel controls and indicators. (continued)

Control/Indicator	Description
Baseline	Filename and path of the ASCII text file containing baseline hearing thresholds for the subject. The baseline file must consist of two columns, the first containing the frequency, the second containing the subject's threshold at each frequency. Each column is separated by a tab; a carriage return is at the end of each row. Baseline thresholds represent the subject's long-term average hearing threshold at each frequency.
Trials to skip	This control specifies how many trials to skip when analyzing a session. When the button to the right of the control is pressed, the number in the numeric field will be added to the second column of the filename list for any selected files. This number of trials will then be skipped when the data are analyzed.
Reversals	The number of reversals to include in the threshold calculations and the number of points within the moving average for the post-exposure thresholds.
Mark as post	Marks the highlighted session as the post-exposure session. The time of the first S1 will be used as the exposure time (exception: if **Manual Exposure Time** is checked, the time entered in the front panel control **Exposure time** will be used instead) and this session will be used for the calculation of the post-exposure thresholds at specific time values.
Exposure time	If the **Options: Manual Exposure Time** menu item is checked, this field acts as a control and is used to enter the time of the exposure. Otherwise, this field shows the exposure time found from the EVT file (the time of the first S1 from the post-exposure session).
Exposure level	May be used to assign a numeric value to the exposure. Used only for record-keeping when exporting session data to MATLAB®.
Filename list	List of SDF files found in the **SDF path** according to the **Filter** specification. The second column indicates how many trials will be skipped in each session when it is analyzed. Individual sessions are selected by shift-clicking on them.
Results	Table showing the results for each session that was selected and analyzed. This table shows the test frequency (Hz), threshold time (in min), threshold SPL and standard deviation, the number of reversals used, the TS (the TS is by definition zero for pre-exposure sessions), the subject's baseline threshold at that frequency (from the **Baseline** file), the threshold relative to the baseline value, the maximum and minimum thresholds (really only makes sense for the post-exposure thresholds), and the difference between the minimum and maximum post-exposure thresholds.
Graphical results	A plot of the subject's thresholds as functions of time. Each frequency is displayed in a different color. The text file colors.txt in the TEXT folder specifies the color mapping. This file is an ASCII, tab-delimited text file containing two columns: the first specifies the frequency, the second indicates the color code for that frequency. Codes are 1 = red, 2 = blue, 3 = green.

7.3 MENU REFERENCE

Tables 26 through 28 list the Compare Multiple Sessions front panel menus

Table 26. Compare Multiple Sessions **File** menu options.

Option	Description
Analyze Selected Files	Causes the selected files to be analyzed and the results displayed.
Save Tabular Results	Saves the tabular results as a tab-delimited ASCII text file.
Save Graphic	Saves the tabular and graphical results as a TIF file.
Export Data	Exports the results to MATLAB®.
Print	Prints the tabular and graphical results.
Close	Close this window and return to the HTP.EXE front panel.

Table 27. Compare Multiple Sessions **Edit** menu options.

Option	Description
Copy Figure to Clipboard	Copies the tabular and graphical results to the Microsoft® Windows clipboard (allows the graph to be pasted into another application).

Table 28. Compare Multiple Sessions **Options** menu.

Option	Description
Manual Exposure Time	If this option is checked, the exposure time is read from the Exposure time front panel control. Otherwise, the exposure time is defined as the time of the first S1 of the post-exposure session.

8. WORKING WITH THE DATALOG FILE

8.1 INTRODUCTION

The datalog file is a database used to store the results from all of the test sessions. The datalog may be exported to spreadsheet applications such as Microsoft® Excel, where results of individual sessions may be extracted using "lookup table" functions. The datalog is a single file containing the complete information regarding the parameters (e.g., frequency, hydrophone settings) and results (e.g., threshold, number of false alarms) from multiple sessions. Figure 35 shows the View Datalog front panel.

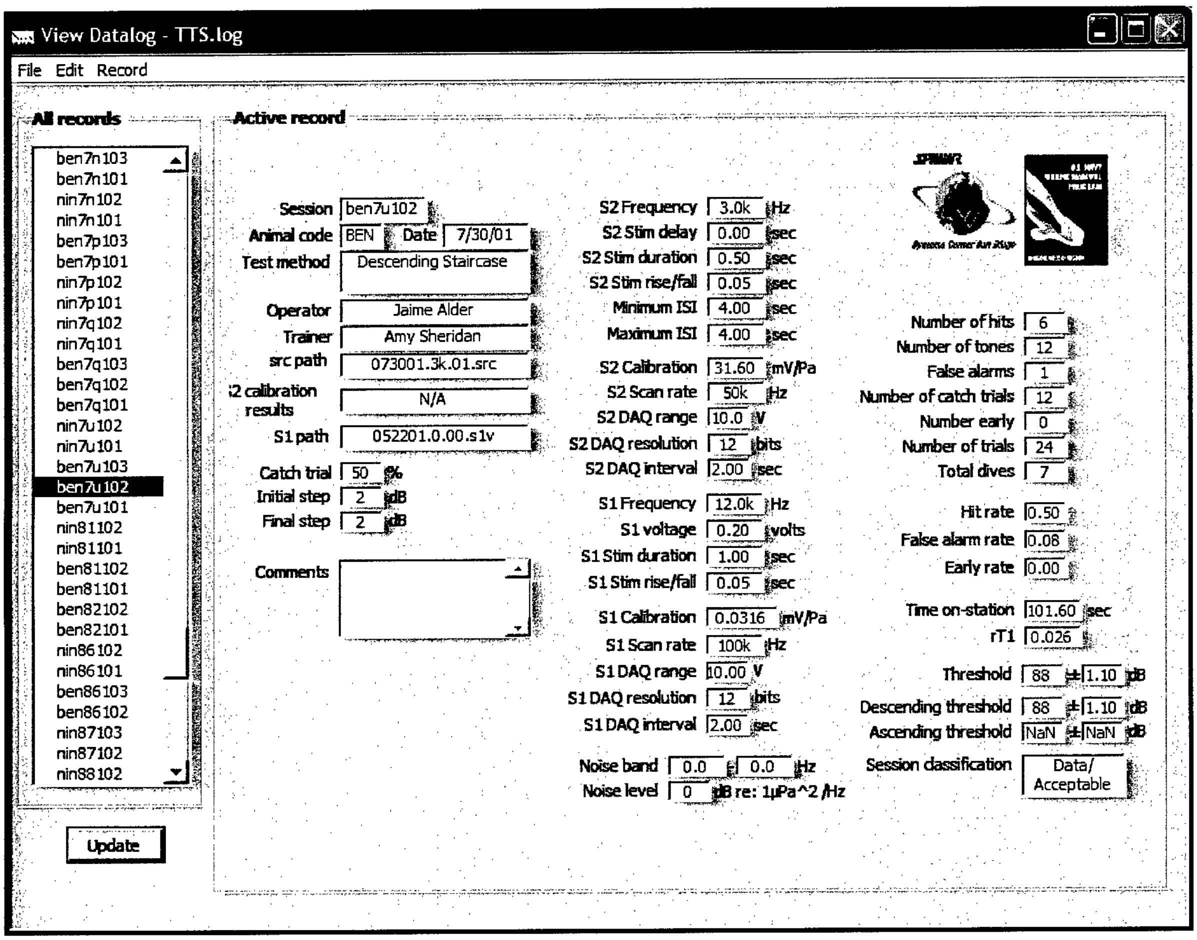


Figure 35. View Datalog front panel.

8.2 DATALOG STRUCTURE

A single datalog record is used to store the information from a single session. Each record contains a number of fields; each field stores a single piece of information, such as the number of false alarms, or the subject's code.

8.3 VIEWING AND MODIFYING THE DATALOG

The datalog may be viewed and/or modified either from the HTP.EXE front panel **File: Open Datalog ...** menu option or the **Analyze Session File: View Datalog ...** menu option.

These menu options open the View Datalog front panel. The left side of this window shows a list of all the sessions in the current datalog (i.e., All records). The right portion of the window shows the datalog fields and their values for the selected session (i.e., the Active record). Values for specific fields may be changed directly in this window; however, if a new record is accessed, all changes will be lost unless the datalog is saved before the new record is accessed. If any changes have been made, the datalog filename will have a '*' appended to it.

8.4 MENU REFERENCE

Tables 29 through 31 describe the various View Datalog menu options.

Table 29. View Datalog **File** menu options.

Option	Description
Open	Open another datalog file.
Save	Save changes to this datalog. If a savable change has taken place, the filename will be appended with an '*'.
Save As	Save the datalog under a new name.
Import	Import ASCII tab-delimited text data into the datalog as new records. Each record of the datalog is a row in the text file. Each column in the text file corresponds to one of the datalog fields.
Export	Export the datalog as an ASCII tab-delimited text file. Each record of the datalog is a row in the text file. Each column in the text file corresponds to one of the datalog fields.
Close	Close this window and return to the HTP.EXE front panel.

Table 30. View Datalog **Edit** menu options.

Option	Description
Cut	Standard Microsoft® Windows Cut command.
Copy	Standard Microsoft® Windows Copy command.
Paste	Standard Microsoft® Windows Paste command.
Clear	Standard Microsoft® Windows Clear command.

Table 31. View Datalog Record menu options.

Option	Description
New	Create a new, blank record.
Duplicate	Duplicate the selected datalog record.
Delete	Delete the selected record.
Find	Search the datalog for a specific session name.

8.5 FILE FORMATS

The datalog file is stored as a LabVIEW™-specific binary file and will not be easily read or written to by other applications.

The exported text files (and any text files that the user intends to import) are ASCII tab-delimited text files. Each row contains the information for a single session, followed by a carriage return. Each column contains the value for a specific field. There are 64 fields stored in the datalog (Table 32). The fields numeric1—numeric5 and string1—string4 are placeholders created in case new parameters are added in the future.

Table 32. Datalog parameters.

Column	Field	Column	Field	Column	Field
1	session	23	s2DAQinterval	45	ER rate
2	subject	24	s1freq	46	stationTime
3	date	25	s1volt	47	rT1
4	method	26	s1stimDuration	48	threshold
5	operator	27	s1stimRise	49	stdev
6	trainer	28	s1mV/Pa	50	descending
7	src	29	s1scanRate	51	stdev
8	s2cal	30	s1DAQrange	52	ascending
9	s1v	31	s1DAQres	53	stdev
10	catch%	32	s1DAQinterval	54	class
11	initialStep	33	HP	55	comments
12	finalStep	34	LP	56	numeric1
13	s2freq	35	noiseLevel	57	numeric2
14	s2stimDelay	36	numHits	58	numeric3
15	s2stimDuration	37	numTones	59	numeric4
16	s2stimRise	38	FA	60	numeric5
17	minISI	39	numCatch	61	string1
18	maxISI	40	numEarly	62	string2

Table 32. Datalog parameters. (continued)

Column	Field	Column	Field	Column	Field
19	s2mV/Pa	41	numTrials	63	string3
20	s2scanRate	42	numDives	64	string4
21	s2DAQrange	43	Hit rate		
22	s2DAQres	44	FA rate		

9. S1 AND FS CALIBRATION

9.1 OVERVIEW

S1 signals and FS tones are calibrated using the S1 and FS Tone Calibration front panel, obtained by selecting the **Calibration: S1 and FS: Tonal** menu option. This front panel (Figure 36) has a tab control and a green **Send S1** button on the left side, two plots in the center, and a table at the bottom. The tab control is used to set various options regarding the sound generation, recording, and analysis. The values in the tab control are retained after closing the S1 and FS Tone Calibration screen (exception: the S1 amplitude is not saved). The upper plot shows the measured time waveforms. The lower plot shows the frequency spectra from the portion of the waveforms inside the two red vertical bars in the upper plot. The table gives the measured SPL, total harmonic distortion, and acoustic energy flux density from each hydrophone channel as well as the mean value of the two channels.

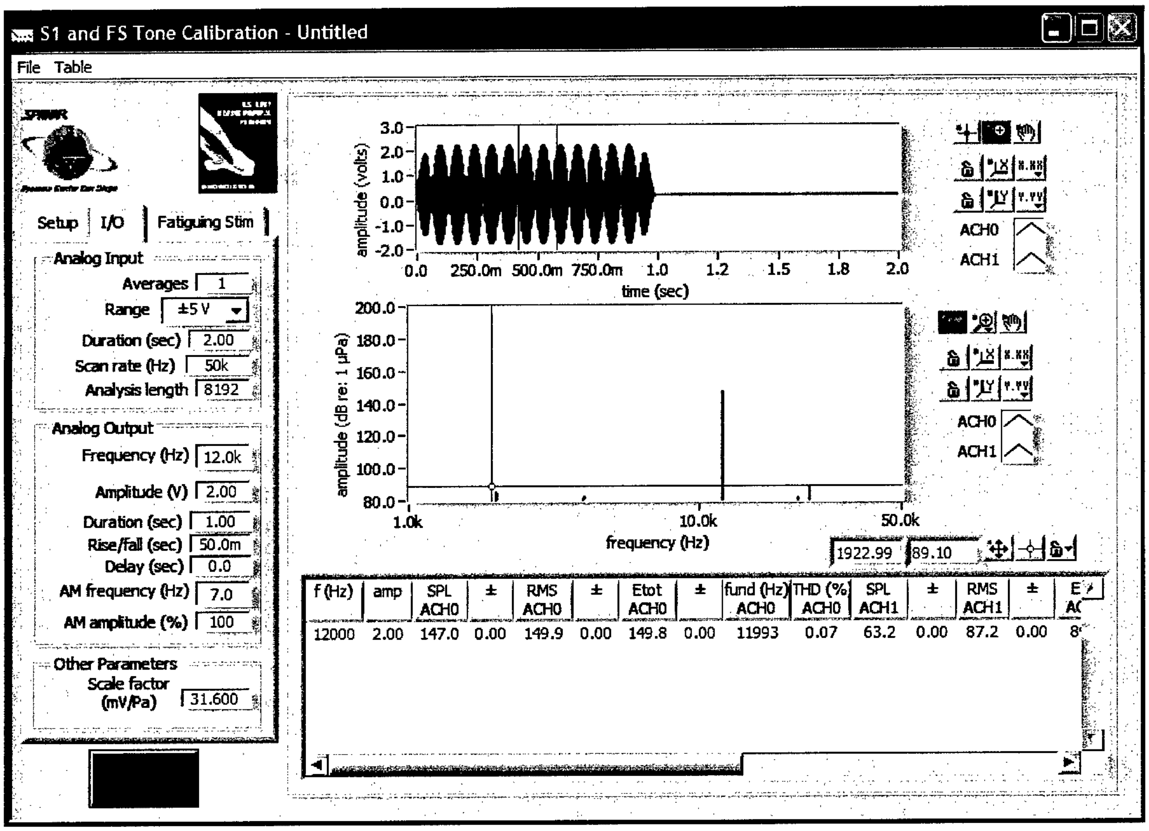


Figure 36. S1 and FS Tone Calibration front panel.

9.2 PROCEDURE

The **Use Fatiguing Stim** checkbox on the Fatiguing Stim tab determines whether the S1 signal or the FS tone is being calibrated. If this box is not checked, then the analysis frequency is taken from the S1 Analog Output **Frequency** control and the position of the analysis window is determined from the S1 **Delay** and **Duration** controls (the analysis window is centered within the S1 signal). If the

Use Fatiguing Stim checkbox is checked, then the FS **Frequency** is used and the analysis window is centered on the FS tone.

9.2.1 S1 Calibration

To calibrate the S1, first check to ensure that the **Use Fatiguing Stim** checkbox is not checked. Check to make sure that the front panel **Scale Factor** value matches the charge amp settings. On the I/O tab, set the **Frequency, Duration, Rise/fall, Delay, Amplitude, AM frequency**, and **AM amplitude** to the desired values (e.g., Analog Output settings: frequency of 12 kHz, duration of 1 s, rise/fall time of 50 ms, delay of 0 s, AM frequency of 7 Hz, and AM amplitude of 100%) or load values using the **File: Open: S1** menu option. Press the **Send S1** button to send an S1 signal. The upper plot will show the measured time waveforms and the lower plot will show the frequency spectra. The frequency spectra should have peaks near the Analog Output **Frequency** that are well above the lower level noise. The table shows the results for both hydrophones (specified in the ACH control) from left to right. Adjust the **Amplitude** control until the ACH1 SPL column indicates the desired level. Save the S1V file using the **File: Save: S1** menu option.

9.2.2 FS Calibration

Check the **Use Fatiguing Stim** checkbox. Check to make sure that the front panel **Scale Factor** value matches the charge amp settings. On the I/O tab, set the **Amplitude** to 0 (this prevents an S1 signal from being generated at the same time). On the Fatiguing Stim tab, set the **Frequency, Duration, Rise/fall, Delay**, and **Amplitude** to the desired values (or load values using the **File: Open: Fatiguing Stimulus** menu option). Set **Averages** to 1.

Press the **Send S1** button to generate the FS and measure the resulting sound in the water. The upper plot will show the measured time waveforms and the lower plot will show the frequency spectra. The frequency spectra should have peaks at the FS frequency; there may also be large peaks at harmonics (integer multiples) of the FS frequency. Adjust the FS **Amplitude** control and/or the FS attenuator and continue to generate and measure the FS tones until the SPL mean column reads the desired value. Save the tabular results using the **Table: Save Table** menu option. Save the FS waveform using the **File: Save: Fatiguing Stimulus** menu option. Print the results using the **File: Print** menu option. Save the individual waveforms using the **File: Save: Waveforms** menu option.

9.3 FRONT PANEL REFERENCE

Tables 33 through 35 describe the S1 and FS Tone Calibration front panel controls and indicators.

Table 33. S1 and FS Tone Calibration **Setup** tab.

Control/Indicator	Description
Devices and Channels	
AI Device	Input device number for S1 and FS hydrophone calibration recordings.
ACH	Analog input channels to record during S1 and FS calibration.
AO Device	Output device number for S1 signals during calibration.
DAC	S1 signal output channel.
Max update rate	Update rate to use when generating the S1 signals.
Base path	Location of the HTP folder on the hard drive.

Table 34. S1 and FS Tone Calibration **I/O** tab.

Control/Indicator	Description
Analog Input	
Averages	Number of averages to use during S1 and FS measurements.
Range	Analog input DAQ range. Should be just larger than the maximum voltage produced by the hydrophone/charge amp.
Duration	Duration of hydrophone recording.
Scan rate	ADC rate for hydrophone signals.
Analysis length	Number of scans to analyze. Must be a power of two.
Analog Output	
Frequency	S1 signal frequency.
Amplitude	S1 signal amplitude.
Duration	S1 signal duration (includes rise and fall).
Rise/fall	S1 signal rise and fall times. A linear rise and fall is used.
Delay	Time delay between hydrophone recording start and generation of S1 signal.
AM frequency	Frequency of S1 signal amplitude modulation.
AM amplitude	Relative amplitude of S1 modulating signal.
Other Parameters	
Scale factor	Hydrophone/charge amp scale factor in mV/Pa.

Table 35. S1 and FS Tone Calibration **Fatiguing Stim** tab.

Control/Indicator	Description
Use Fatiguing Stim	Check box to determine if a FS is sent. If **Use Fatiguing Stim** is checked, then the analysis window is centered within the FS and the analysis uses the FS frequency; otherwise, the S1 parameters are used.
Analog Output	
Frequency	FS tone frequency.
Amplitude	FS tone amplitude.
Duration	FS tone duration (includes rise and fall).
Rise/fall	FS tone rise and fall times. A linear rise and fall is used.
Delay	Time delay between hydrophone recording start and generation of FS tone.
Device and DAC	
AO Device	Output device number for FS tone.
DAC	Output channel for FS tone.
Update rate	Update rate to use when generating the FS tone.

9.4 MENU REFERENCE

Tables 36 and 37 list the S1 and FS Tone Calibration menu options.

Table 36. S1 and FS Tone Calibration **File** menu.

Option	Description
Open	
S1	Open an existing S1V file and change current S1 parameters to match those in the file.
Waveforms	Open hydrophone recordings saved as a binary file (i16, Intel format, 2-channel, interleaved).
Save	
S1	Save the current parameters necessary to generate the S1 signal. File is saved as an S1V file.
Fatiguing Stimulus	Save the current FS waveform as a binary file (signed i16, Motorola format, 1-channel).
Waveforms	Save the hydrophone recordings as a binary file (signed i16, Intel format, 2-channel, interleaved).
Print	Print the calibration results.
Close	Close this window and return to the HTP.EXE front panel.

Table 37. S1 and FS Tone Calibration **Table** menu.

Option	Description
Save Table	Save the tabular results as an ASCII tab-delimited text file.
Clear All Rows	Erase all rows in the table.
Clear Selected Row(s)	Erase any selected rows in the table. Use shift-click to select more than one row.

9.5 DATA FILE FORMATS

9.5.1 S1V Files

S1V ("S1 voltage") files are created during the S1 calibration process. These files store the information necessary to create the S1 signal. S1V files are ASCII tab-delimited text files consisting of a single column of seven numbers indicating the S1 frequency, amplitude (i.e., voltage), duration (in seconds), rise and fall times (in seconds), minimum delay (in seconds) from S1 send button to actual sound, modulation (AM) frequency (in Hz), and modulation amplitude (%).

9.5.2 FS Files

FS ("fatiguing stimulus") files store the FS waveform defined during the FS calibration process. FS files are single-channel, 16-bit binary files (Motorola format).

10. S2 CALIBRATION

10.1 OVERVIEW

Accurate estimates of a subject's hearing threshold rely upon precise knowledge of the sound pressure produced at the subject's location. This estimate is achieved by *calibrating* the S2 tones so that the user knows exactly how much voltage to generate in order to produce an S2 tone with a specific SPL. The S2 calibration has three main steps: (1) adjust the S2 Power Amp volume and the S2 Attenuator, (2) measure the acoustic pressure as a function of the computer output (DAQ) voltage, and (3) measure the actual SPL versus the desired SPL to verify the calibration.

10.1.1 S2 Power Amp volume and S2 Attenuator Adjustment

The first step in the S2 calibration is to set the S2 Power Amp and S2 Attenuator at the appropriate values. The objective of this step is to set the amp and attenuator to allow the computer generated voltages (which are restricted to the range of ±10 V) to produce SPLs ranging from the user-defined **Low SPL** and **High SPL.** This objective is achieved by fixing the amp volume and then adjusting the S2 Attenuator to put the measured SPL near the **High SPL** value when outputting 10 V (the maximum output voltage).

10.1.2 Pressure vs. DAQ Voltage Measurement

The next calibration step is to measure the sound pressure produced when specific voltages are generated by the PC and use these values to calculate the calibration constant (in Pa/V) for the system. The measurement proceeds as follows: The computer outputs a number of discrete voltages ranging from the front panel **Low volts** to **High volts** in a number of steps indicated by **Divisions,** then reverses the sequence (i.e., from **High volts** to **Low volts** in **Divisions** number of steps). If **log** is checked, the spacing between voltages is logarithmic and not equal. This process provides fine detail near 0 V without requiring a large number of **Divisions.** At each voltage, the acoustic pressure is measured and displayed in the upper left plot. From this measurement, the computer calculates the voltages required to generate individual SPLs ranging from **Low SPL** to **High SPL** in 1-dB increments. These values are displayed in a table which the user is prompted to save as an SRC (source) file. If the file is saved, the procedure automatically continues to the next step: Actual SPL vs. Desired SPL measurement.

The success of this step is ascertained by examining the data points within and the $\mathbf{R^2}$ and mean squared error (**mse**) values beneath the upper left plot. The $\mathbf{R^2}$ value should be at least 0.99 and the mse should be roughly 20×10^{-3} or lower.

10.1.3 Actual SPL vs. Desired SPL Measurement

During this step the computer outputs the voltages from the SRC file previously saved (these voltages correspond to those calculated to produce SPLs ranging from **Low SPL** to **High SPL**) and measures the SPL at each voltage. These data are displayed in the upper right plot as the measured SPL vs. the desired SPL.

The success of this step is evaluated by examining the $\mathbf{R^2}$ and correlation coefficient ($\rho_{x,y}$) values displayed beneath the lower right plot: the $\mathbf{R^2}$ and $\rho_{x,y}$, values should be at least 0.99.

At the conclusion of the Actual SPL vs. Desired SPL measurement, the program automatically saves the calibration results in a MAT file located in the S2CAL folder. The MAT file has a filename derived from the SRC filename previously defined by the user.

10.2 PROCEDURE

10.2.1 Begin

To begin the S2 calibration, select the **Calibrate: S2** menu item.

10.2.2 Check Front Panel Settings

Check the current values for the various front panel variables listed along the left side of the screen. Most of the default values will generally be appropriate. **Frequency**, **High SPL**, and **Low SPL** will be defined before testing.

10.2.3 Set S2 Amp Volume and S2 Attenuator

The objective of this step is to set the S2 Amplifier and S2 Attenuator to appropriate values to achieve the desired range of SPLs. From the S2 Calibration **Manual** menu, select **Start**.

Set the S2 Power Amp volume to an appropriate setting (normally, the Power Amp volume is kept fixed from day to day). Set the I/O tab **Amplitude** to 10.00 V. Send an S2 tone by clicking the **Send S2** button. Use the scroll bar to view the last measurement result on the table (if necessary). This shows the average SPL at the hydrophone(s). Adjust the S2 Attenuator and send additional S2s until the measured average SPL is close to the desired value, normally 0.5 to 1 dB above the **High SPL** setting. *Note that the S2 Attenuator should never be set to less than 10.0 dB.*

Now is also a good time to verify that the **Low volts** setting is high enough to produce a clear peak in the frequency spectrum at the test **Frequency**, i.e., **Low volts** produces a sound pressure that is above the ambient noise. To do this, set the **Amplitude** equal to **Low volts** and press **Send S2.** Look at the frequency spectrum plot (the lower of the two) and verify that a clear peak exists at the test frequency. If not, increase the **Amplitude** and send another S2 (and repeat as necessary) to find the lowest **Amplitude** at which a clear peak is visible above the ambient noise and use that in place of the previous **Low volts**.

When finished, select **Close** from the **File** menu and return to the main S2 Calibration front panel.

10.2.4 Automatic Calibration

The next three steps of the S2 Calibration are performed using the Automatic Calibration menu option. To begin, select **Start Full Calibration** from the **Automatic** menu.

10.2.4.1 Pressure vs. DAQ Voltage

This measurement will begin automatically after selecting **Start Full Calibration**. The computer outputs a number of voltages ranging from the front panel **Low volts** to **High volts** in a number of steps indicated by **Divisions**, then reverses the sequence (i.e., from **High volts** to **Low volts** in **Divisions** number of steps). If **log** is checked, the spacing between voltages is logarithmic rather than linear. This display provides fine detail near 0-V without requiring a large number of **Divisions**. At each voltage, the acoustic pressure is measured and displayed in the upper left plot. During each measurement a window will pop-up on the right side of the screen showing the measured time waveforms and frequency spectra (Figure 37). Frequency spectra should show a clear peak at the test frequency.

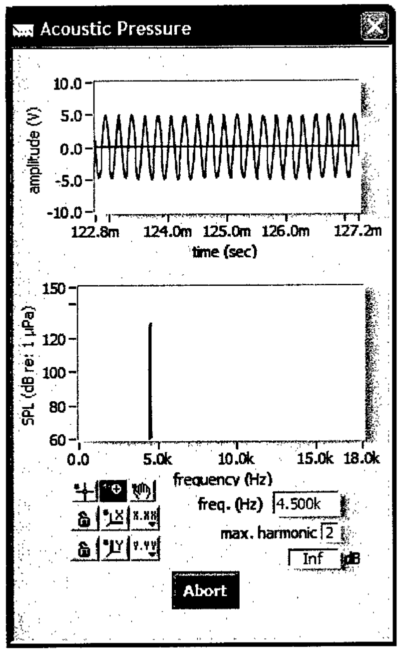


Figure 37. Popup window showing pressure time waveform (upper) and frequency spectrum (lower). A clear peak should exist in the frequency spectrum at the S2 frequency.

The computer uses the measured data to generate a polynomial describing the system transmitting response relating pressure and voltage. The polynomial has the form

$$p(v) = a_2 v^2 + a_1 v + a_0,$$

where p is the acoustic pressure (in Pa), v is the DAQ voltage (in volts), and a_2, a_1, and a_0 are constants. If the I/O tab **poly fit** box is not checked, the constant, a_2, is set equal to zero (a linear fit is used, rather than polynomial). From the equation, the computer calculates the voltages required to generate individual SPLs ranging from **Low SPL** to **High SPL** in 1-dB increments. These values are displayed in a table at the bottom center of the screen. If the full range of SPLs from **Low SPL** to **High SPL** can be generated using the available computer output voltage range (approximately 0.1–10 V), then the user is prompted to save these data as an SRC (source) file. When saving SRC files, be sure to use the correct filename format: mmddyy.f.nn.src, where mm is a two-digit month, dd is a two-digit day, yy is a two-digit year, f is the frequency (e.g., 1000 Hz = "1k", 4.5 kHz = "4_5k"), and nn is a two-digit number starting with 00 for the first saved SRC of the day. For example, the first SRC file saved on October 15, 1999 at 10.5 kHz would be named 101599.10_5k.00.src.

If the **High SPL** cannot be generated without exceeding **Max Volts**, then an Out of Range Error is reported. If this happens, press OK to clear the error message, lower the S2 Attenuator setting or increase the S2 Amp volume, then re-select **Start Full Calibration** from the **Automatic** menu. *Remember that the S2 Attenuator should never be set to less than 10.0 dB.*

The success of the pressure vs. DAQ voltage measurement is determined by examining the $\mathbf{R^2}$ and **mse** values beneath the upper left plot ($\mathbf{R^2}$ is a correlation coefficient and would be 1.0 for a perfect match between the experimentally measured pressure and the least squares curve-fit; **mse** is the mean-squared-error and would be zero for perfect agreement between the measured pressure and the polynomial approximation). The **R^2** value should be at least 0.99. If the data do not meet this criterion, press **Cancel** and then start a new automatic calibration by selecting **Start Full Calibration** from the **Automatic** menu. Continue to repeat until the data meet the success criterion. If a large amount of scatter is observed, the user may increase the **Averages** to help reduce scatter caused by measurement variability.

Once the file is saved (i.e., the data are acceptable), the procedure automatically moves forward to the next step: Actual SPL vs. Desired SPL measurement.

10.2.4.2 Actual SPL vs. Desired SPL

During this step, the computer outputs the voltages from the SRC file previously saved (these voltages correspond to those calculated to produce SPLs ranging from **Low SPL** to **High SPL**) and measures the SPL at each voltage. These data are displayed in the upper right plot as the measured SPL vs. the desired SPL. The lower right plot shows the error, defined as the difference between the measured and desired SPLs, vs. the desired SPL. Again, during each measurement, a window will pop-up on the right side of the screen showing the measured time waveforms (top) and frequency spectra (bottom). Frequency spectra should show a clear peak at the test frequency; however, at low SPLs, the measured pressure will be buried in the ambient noise, so a clear peak may not be visible until the SPL reaches 80 to 90 dB re 1 μPa.

The measured data are plotted (blue symbols) along with the desired curve (red line), a line with a slope of 1.0 and an intercept of 0.0. At the conclusion of the measurement, the user is asked if the data are acceptable. The success of the measurement is evaluated by examining the **R^2** and $\rho_{x,y}$ values displayed beneath the upper right plot. The data will exhibit some degree of scatter at the lower levels, but should eventually merge and closely follow the desired curve (red line). The **R^2** value should be at least 0.99. If the data do not meet this criterion, press No to indicate that the data are not acceptable and a new measurement will automatically begin. Repeat the measurements until the criterion above is met, then press Yes to indicate that the data are acceptable. The program will save the results.

10.2.5 Finishing Up

At the conclusion of the Actual SPL vs. desired SPL measurement, the program also automatically saves the calibration results in the S2CAL folder with a filename derived from the SRC filename previously defined by the user. For example, if the SRC filename was 101599.1k.00.src, then the calibration results are saved as s2cal101599.1k.00.mat. It is important to follow the naming conventions and not rename files—during data analysis some calibration files are automatically found by the program based on the SRC filename—it is important that they match.

At this point the user should print out the S2 calibration data by selecting **Print** from the **File** menu. The S2 Calibration front panel may then be closed by selecting **Close** from the **File** menu. This action returns the user to the main HTP.EXE front panel.

10.3 FRONT PANEL REFERENCE

Figure 38 shows the S2 Calibration front panel. Tables 38 and 39 describe the various front panel controls and indicators.

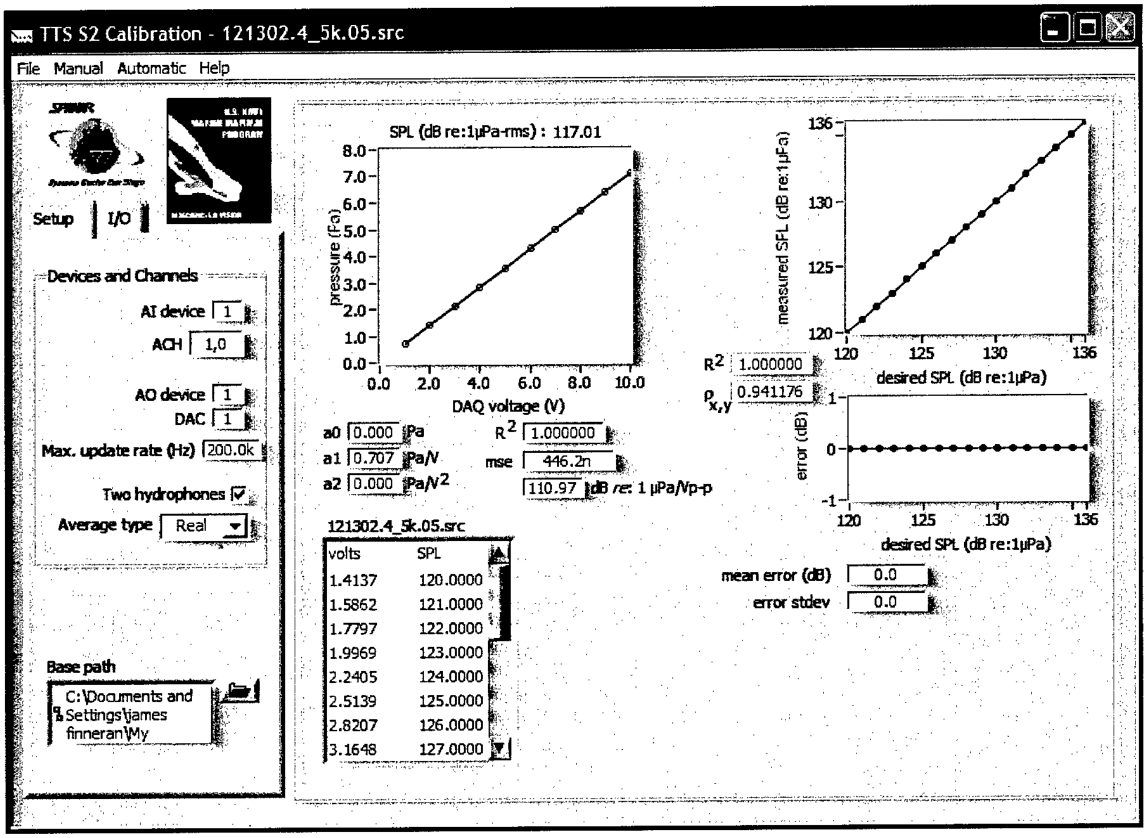


Figure 38. S2 Calibration front panel.

Table 38. S2 Calibration **Setup** tab.

Option	Description
Devices and channels	
AI Device	Input device number for S2 hydrophone calibration recordings.
ACH	Analog input channels. Multiple channels are separated by commas. If **Two hydrophones** is not checked, only the first channel listed will be used.
AO Device	Output device number for S2 signals during calibration.
DAC	S2 signal output channel.
Max update rate	Update rate to use when generating the S2 signals.
Two hydrophones	If this box is checked, recordings will be made from both hydrophones and the average SPL used. Otherwise, the SPL from the first channel listed in **ACH** will be used.

Table 38. S2 Calibration **Setup** tab. (continued)

Option	Description
Average type	Method of averaging SPLs from two hydrophones. Use **Complex** average type to include phase information in the average. **Real** average does not include phase information.
Base path	Location of the HTP folder on the hard drive.

Table 39. S2 Calibration **I/O** tab.

Option	Description
Analog Input	
Averages	Number of averages to use during the S2 measurements.
Range	Analog input DAQ range. Should be just larger than the maximum voltage produced by the hydrophone/charge amp.
Duration	Duration of hydrophone recording.
Scan rate	ADC conversion rate for hydrophone signals.
Analysis length	Number of scans to analyze. Must be a power of two.
Analog Output	
Frequency	S2 tone frequency.
Duration	S2 signal duration (include rise and fall).
Rise/fall	S2 signal rise and fall times. A linear rise and fall is used.
Delay	Time delay between hydrophone recording start and generation of S2 tones.
Max volts	Maximum voltage allowed to be generated by the computer. If the voltage necessary for a particular SPL is greater than **Max volts**, an Out of Range Error will occur.
Low volts	Starting voltage for the Pressure vs. DAQ voltage measurements. Using this voltage on the S2 Manual Calibration should result in an SPL above ambient noise. This setting also controls the lowest SPL at which the correlation measurements and R^2 values will be computed (indicated by the starting point of the red line).
High volts	Stopping voltage for the Pressure vs. DAQ voltage measurements. Must be less than or equal to Max volts.
Divisions	Number of individual measurements to make between **Low volts** and **High volts** during the Pressure vs. DAQ voltage measurements. Increasing this number results in a better curve-fit, but increased time for calibration.

Table 39. S2 Calibration **I/O** tab. (continued)

Option	Description
log	If this box is checked, a log spacing (rather than a linear spacing) will be used to determine the actual voltages to send out during the Pressure vs. DAQ voltage measurements. Checking the **log** box results in relatively more measurements being made at the lower voltage settings. For example: If **Low volts** = 1, **High volts** = 10, **Divisions** = 5 and **log** is not checked, the voltages used will be 1, 3.2, 5.5, 7.8, 10; if **log** is checked, the voltages will be 1, 1.8, 3.2, 5.6, 10.
Low SPL	The lowest SPL to use in the actual SPL vs. desired SPL measurements. This SPL is the lowest S2 SPL that will be available during a test session.
High SPL	The highest SPL to use in the actual SPL vs. desired SPL measurements. This SPL is the highest S2 SPL that will be available during a test session.
Other Parameters	
Scale factor	Hydrophone/charge amp scale factor in mV/Pa.
poly fit	If this box is checked, a polynomial of the form $p(v) = a_2v^2 + a_1v + a_0$ will be used to fit the pressure vs. DAQ voltage data. Otherwise, an equation of the form $p(v) = a_1v + a_0$ will be used.

10.4 MENU REFERENCE

Tables 40 through 43 list the S2 Calibration menu options.

Table 40. S2 Calibration **File** menu.

Option	Description
Open	Open an existing s2cal file. This option allows the results of a previous calibration to be viewed/printed.
Print	Print the results of the calibration.
Close	Close this window and return the HTP.EXE front panel.

Table 41. S2 Calibration **Manual** menu.

Option	Description
Start	Open the S2 Manual calibration front panel.

Table 42. S2 Calibration **Automatic** menu.

Option	Description
Start Full Calibration	Start the full calibration, consisting of the pressure vs. DAQ voltage measurements and the Actual SPL vs. Desired SPL measurements.
Check Existing SRC	Open an existing SRC file and use the voltage/SPL values contained to repeat the Actual SPL vs. Desired SPL measurements. This option is used primarily to check an existing calibration before or after a session.
Map Acoustic Field	Open a new front panel used to map the spatial distribution of the acoustic pressure.

Table 43. S2 Calibration **Help** menu.

Option	Description
Online Reference	Open a PDF version of this document.

10.5 DATA FILE FORMATS

10.5.1 SRC Files

SRC ("source") files are created during the S2 calibration process. The computer uses these files to translate a desired SPL into a specific voltage to be output at the DAC. SRC files are ASCII tab-delimited text files. The first row indicates the frequency. The remaining rows consist of pairs of numbers separated by a TAB. The first number indicates the voltage, the second number gives the SPL produced from that voltage.

11. NOISE CALIBRATION

11.1 OVERVIEW

It is sometimes of interest to measure hearing thresholds in the presence of intentionally generated "noise" with a known amplitude and frequency content. Hearing thresholds measured at a particular frequency may be affected, or "masked," by noise at similar frequencies; therefore, this type of noise is often called "masking noise." Intentionally generated masking noise may be used to provide a consistent background in the presence of more variable noise (beyond the control of the experimenter) or to investigate some particular function of the auditory system, such as the ability of the subject to detect a sound in the presence of noise.

To begin the Masking Noise calibration, select **Noise ...** from the HTP.EXE front panel **Calibration** menu. Masking noise calibration has five steps: (1) Build a digital filter (i.e., not a "real" hardware filter, but one created in software) to generate band-limited (flat) white noise, (2) Measure the acoustic pressure produced when driving the noise projector with the band-limited noise, (3) Build a new digital filter to compensate for the actual appearance of the measured noise (i.e., to make it flat), (4) Measure the acoustic pressure produced when driving the noise projector with the compensated band-limited noise, and (5) Adjust the noise amplitude to obtain the desired noise level.

If the noise projector (and all the other associated electronics, filters, amplifiers, etc.) possessed a perfect transmitting response and the measurements were conducted in a very large body of water (free from any reflective surfaces), the masking noise calibration would consist of generating "flat" noise (i.e., equal amplitude at every frequency) over the desired frequency range and adjusting the measured sound level to the desired level. However, the noise projector's frequency-dependent transmitting response and the presence of multiple reflecting surfaces combine to enhance the sound pressure at the hydrophones for certain frequencies and reduce the pressure at other frequencies, making the measured noise non-flat versus frequency.

To overcome this problem, we generate flat white noise and measure what the acoustic pressure actually looks like. This measurement indicates how much the initially flat noise is distorted by the projector and other electronics and the sound propagation from the noise source to the hydrophone. To compensate for these effects, we generate a digital filter whose shape is the inverse or reciprocal of the measured noise; the filter shape may be thought of as the measured noise flipped upside-down: if the measured noise has a peak at a certain frequency, we must reduce the amplitude of the (initially flat) noise at that frequency so that the actual sound pressure will end up flat. Flat noise is then played through this digital compensation filter before being amplified and applied to the noise projector. In theory, the digital filter will perfectly compensate for the projector transmitting response and the effects of the sound water path from the source to the received hydrophones and the resulting measured noise will be flat.

In practice, many factors will make the measured noise less than perfectly flat. The pressure measurements are based on the pressure at one hydrophone. Any changes in the water temperature or depth (e.g., due to tides) will change the results, as will the presence of small fish moving through the enclosure. Any large ambient noise present during the initial measurement will give a distorted view of the effect of the projector and water path on the initially flat noise and will cause the compensated noise to be non-flat.

11.2 PROCEDURE

11.2.1 Check Front Panel Settings

Check the current values for the various front panel variables listed along the left side of the screen. The **Analysis length** must be a power of 2 (e.g., 512, 1024, 2048, 4096, 8192). Decreasing **Analysis length** will lower the frequency resolution (the ability to resolve closely spaced frequencies), which results in a more "coarse" measurement. It may be necessary to decrease **Analysis length** if the presence of sharp peaks in the measured spectrum causes problems when building compensation filters. Increasing **Analysis length** improves the frequency resolution. The **Scan rate** must be greater than twice the maximum frequency of interest. Increasing the **Scan rate** will decrease the frequency resolution (the opposite effect of the **Analysis length**). If the **Scan rate** must be increased, one may find it necessary to also increase the **Analysis length** to maintain a useable frequency resolution. Figure 39 shows the Masking Noise Calibration front panel.

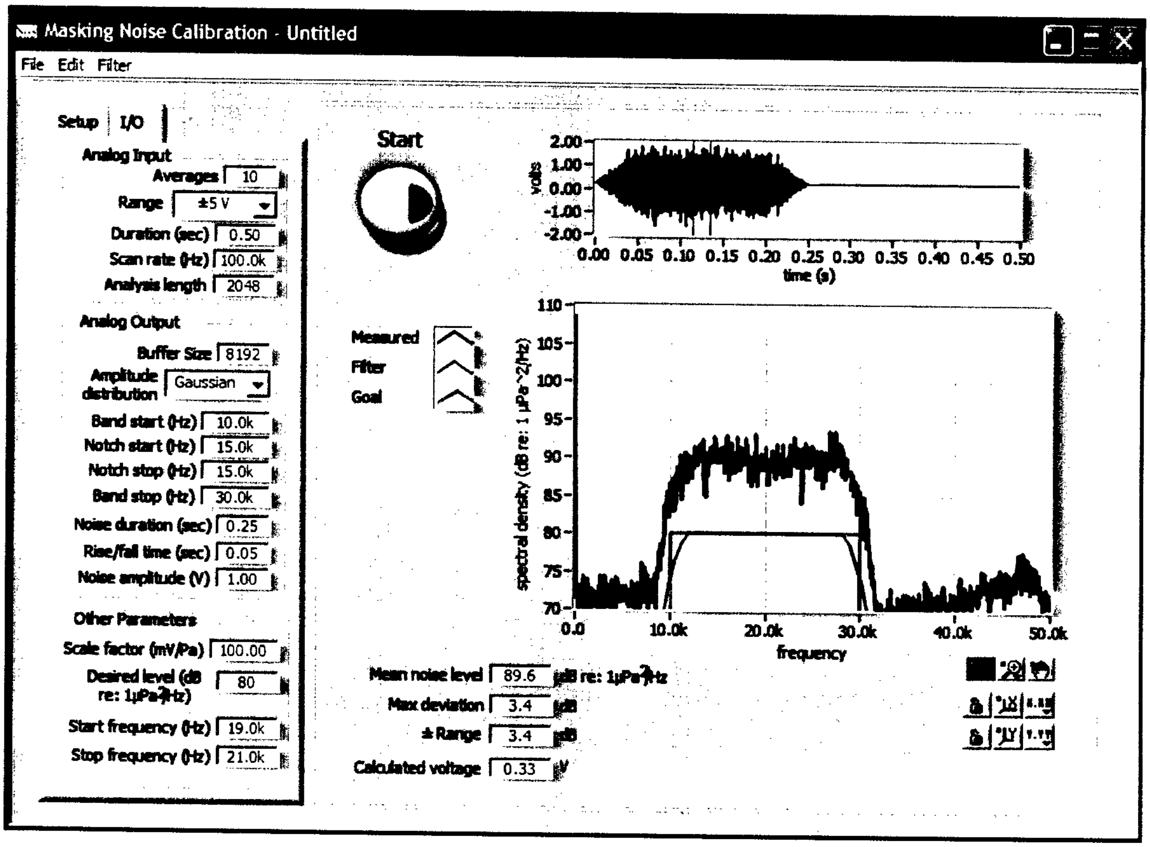


Figure 39. Masking Noise Calibration front panel.

11.2.2 Define Noise Parameters

The first step in calibrating the masking noise is to define the frequency band over which the noise is desired. The front panel controls **Band start, Band stop, Notch start,** and **Notch stop** (Figure 40) are used for this purpose. **Band start** and **Band stop** define the total frequency range over which the noise will be generated. **Notch start** and **Notch stop** are used to define a region of no-noise (i.e., a spectral "notch" within the noise) that may be placed within the noise band. If no notch is desired, set **Notch start** equal to **Notch stop**. Note that **Band start** < **Notch start** <= **Notch stop** < **Band stop**. The green lines in the spectral density plot will update to reflect changes made to the four noise bandwidth parameters.

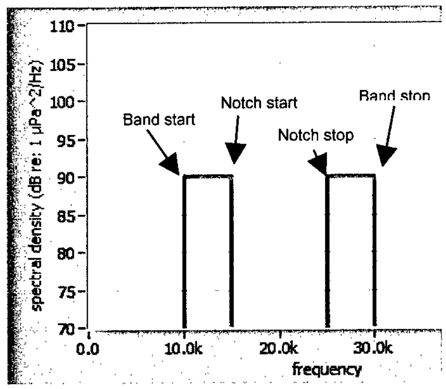


Figure 40. Illustration of **Band start**, **Band stop**, **Notch start**, and **Notch stop** controls.

11.2.3 Build Initial Digital Filter

Before the noise can be generated, a digital filter must be created. This filter tells the computer to only generate noise using the frequency bandwidth parameters that have been defined (i.e., the green lines in the plot). To build the filter, select the **Filter: Build New Filter** menu option. Be sure that the **Use Measured Spectrum** option is not checked. The Build Compensation Filter window (Figure 41) shows the desired filter shape (the red lines) and the shape of the digital filter (blue lines). The **Number of points, Filter Order**, and **Window** controls may be used to adjust the shape of the digital filter, if desired (default values are normally ok). If changes are made, use the **Build** button to build a new filter and plot the results. When finished, press the **OK** button. After choosing OK, the spectral density plot on the Noise Calibration front panel will be updated with the shape of the digital filter that was just created (red lines).

Note that when using notched noise, the initial digital filter should be created using noise without a notch. This action will improve the compensation of the noise within each of the passbands. Also, the noise compensation at the low and high frequency extremes will be improved if the initial noise bandwidth is set to a larger value than the final desired limit, i.e., set **Band start** below the desired low frequency and **Band stop** above the desired high frequency when doing the initial noise filter build.

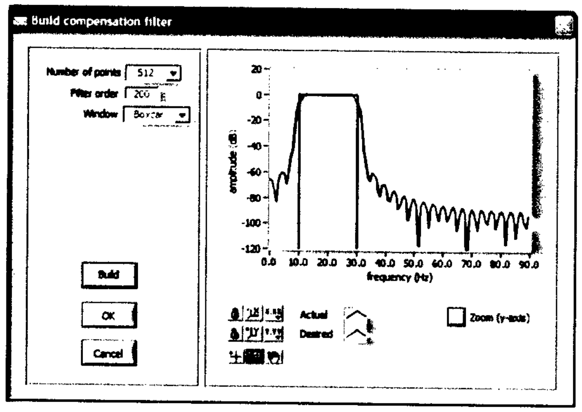


Figure 41. Build Compensation Filter window.

11.2.4 Measure Band-Limited White Noise

After the digital filter is built, press the front panel **Start** button to generate the noise and measure the resulting acoustic pressure.

11.2.5 Build New Compensation Filter

The noise sent to the projector has a uniform frequency content; however, the measured pressure will not be flat. To get the measured pressure to be flat, we pass the computer-generated, band-limited white noise (which *is* flat) noise through a digital filter before it is amplified and input to the sound projector. The shape of this digital filter is derived from the measured pressure obtained when driving the source with band-limited white noise. This concept is sometimes referred to as "pre-equalizing"—we are enhancing some frequencies and reducing others under the expectation that the source transmitting properties and the propagation through the water will tend to cause the opposite effects at these frequencies.

To build the compensation filter, first measure the band-limited white noise, as described above. Then, from the **Filter** menu, select **Build New Filter**. The **Use Measured Spectrum** option must be checked—dictating that the filter being built will use the existing measurement to define its shape.

The Build Compensation Filter window will show the desired filter shape (the red lines) and the shape of the digital filter (blue lines). The **Number of points**, **Filter Order**, and **Window** controls may be used to adjust the shape of the digital filter, if desired (default values are normally OK). If changes are made, use the **Build** button to build a new filter and plot the results. When finished, press the **OK** button. After choosing **OK**, the spectral density plot on the main Noise Calibration panel will be updated with the shape of the digital filter that was just created (red lines).

11.2.6 Generate Compensated Band-Limited Noise

Press **START**. The computer will begin generating noise and measuring the acoustic pressure. Use the **Start frequency** and **Stop frequency** controls to define the frequency range over which to calculate the average noise (spectral density) level. The **Mean noise level**, **Max deviation**, and **±Range** indicators show the descriptive statistics for the measured noise over this frequency range. The **Calculated voltage** indicator shows the required **Amplitude** setting to get the **Mean noise level** to equal the **Desired level**.

11.2.7 Finishing Up

Once an acceptable pressure measurement is obtained, select **Save As** from the **File** menu. The file created is of the type "FIR," which stands for "finite impulse response," a type of digital filter. The FIR files must be saved in the FIR folder. Use the **File** menu **Print** option to print the calibration results.

11.3 FRONT PANEL REFERENCE

Tables 44 and 45 describe the Masking Noise Calibration front panel controls and indicators.

Table 44. Masking Noise Calibration **Setup** tab.

Control/Indicator	Description
Devices and channels	
AI Device	Input device number for noise hydrophone calibration recordings.
ACH	Analog input channel for hydrophone.
AO Device	Output device number for noise during calibration.
DAC	Noise signal output channel.
Update rate	Update rate to use when generating the noise. If a single DAQ device is used, the **Update rate** must be equal to the **S2 update rate**.
Base path	Location of the HTP folder on the hard drive.

Table 45. Masking Noise Calibration **I/O** tab.

Option	Description
Analog Input	
Averages	Number of averages to use during the noise measurements.
Range	Analog input DAQ range. Should be just larger than the maximum voltage produced by the hydrophone/charge amp.
Duration	Duration of hydrophone recording.
Scan rate	ADC conversion rate for hydrophone signals.
Analysis length	Number of scans to analyze. Must be a power of two.
Analog Output	
Buffer size	Analog output buffer size in number of scans (must be a power of two). This control is for compatibility with a separate program used to generate continuous noise.

Table 45. Masking Noise Calibration **I/O** tab. (continued)

Option	Description
Amplitude distribution	Statistical distribution of the noise amplitude. This pull-down menu has two options: Uniform distribution or Gaussian (Normal) distribution.
Band start	Low-cutoff frequency for the noise.
Notch start	Lower frequency for a spectral notch within the noise.
Notch stop	Upper frequency for a spectral notch within the noise. If a notch is not desired, set Notch stop = Notch start.
Band stop	High-cutoff frequency for the noise.
Noise duration	Noise duration (include rise and fall).
Rise/fall	Noise rise and fall times. A linear rise and fall is used.
Noise amplitude	Maximum noise amplitude for Uniform distribution or standard deviation for Gaussian distribution. Note that for Gaussian noise, the maximum ± voltage excursions will be roughly three times the standard deviation. Since the DAQ boards can only produce voltages over the range ±10 V, the **Noise amplitude** should be limited to a maximum of 3 V for Gaussian noise.
Other Parameters	
Scale factor	Hydrophone/charge amp scale factor in mV/Pa.
Desired level	Desired noise level in dB re 1μPa^2/Hz.
Start frequency	Lower frequency bound for analysis range.
Stop frequency	Upper frequency bound for analysis range.

11.4 MENU REFERENCE

Tables 46 through 48 list the Masking Noise Calibration menu options.

Table 46. Masking Noise Calibration **File** menu.

Option	Description
Open	Open an existing FIR file. This option allows the results of a previous calibration to be viewed/printed.
Save As	Save the results of the current calibration to a new FIR file.
Print	Print the results of the calibration.
Close	Close this window and return the HTP.EXE front panel.

Table 47. Masking Noise Calibration **Edit** menu.

Option	Description
Copy	Standard Windows Copy command.
Paste	Standard Windows Paste command.

Table 48. Masking Noise Calibration **Filter** menu.

Option	Description
Build New Filter ...	Opens the Build Compensation Filter front panel. Creates a digital filter using the specified parameters. If **Use Measured Spectrum** is checked, then the filter's response in the passbands will be derived from the inverse of the measured pressure in this region; otherwise, the passband amplitude will be flat.
Use Measured Spectrum	If **Use Measured Spectrum** is checked, then the filter's response in the passbands will be derived from the inverse of the measured pressure in this region, otherwise the passband amplitude will be flat.

11.5 DATA FILE FORMATS

11.5.1 FIR Files

FIR files are created during the noise calibration process. These files are used by the computer to generate masking noise with specific spectral parameters. FIR files are ASCII files that contain detailed information regarding the noise frequency content, amplitude, duration, calibration results, and certain front panel settings.

The first row of the FIR file stores the following front panel parameters listed in Table 49.

Table 49. FIR file header format.

Column	Parameter	Column	Parameter
1	Update rate	11	Noise duration
2	Noise amplitude	12	Start frequency
3	Band start	13	Stop frequency
4	Notch start	14	Format code (2 = HTP program)
5	Notch stop	15	Desired noise level
6	Band stop	16	Plot x-minimum
7	Buffer size	17	Plot x-maximum
8	Amplitude distribution	18	Plot y-minimum
9	Rise/fall time	19	Plot y-maximum
10	N/A		

The second row contains the digital filter coefficients. The third and fourth rows contain the last measured pressure spectrum frequency and amplitude, respectively.

12. IMPULSE CALIBRATION

12.1 OVERVIEW

Impulsive sound sources may be calibrated using the S1 and FS Impulse Calibration front panel (Figure 42), which is obtained by selecting the **Calibrate: S1 and FS: Impulsive** menu option. This front panel has a tab control and a green **Send S1** button on the left side, two plots in the center, and a table at the bottom. The tab control is used to set various options regarding the sound generation, recording, and analysis. The values in the tab control are retained after closing the window. The upper plot shows the measured time waveforms. The lower plot shows the frequency spectra from the portion of the waveforms inside the two red vertical bars in the upper plot. The table gives the measured peak pressure, SPL, acoustic energy flux, and signal duration from both hydrophone channels.

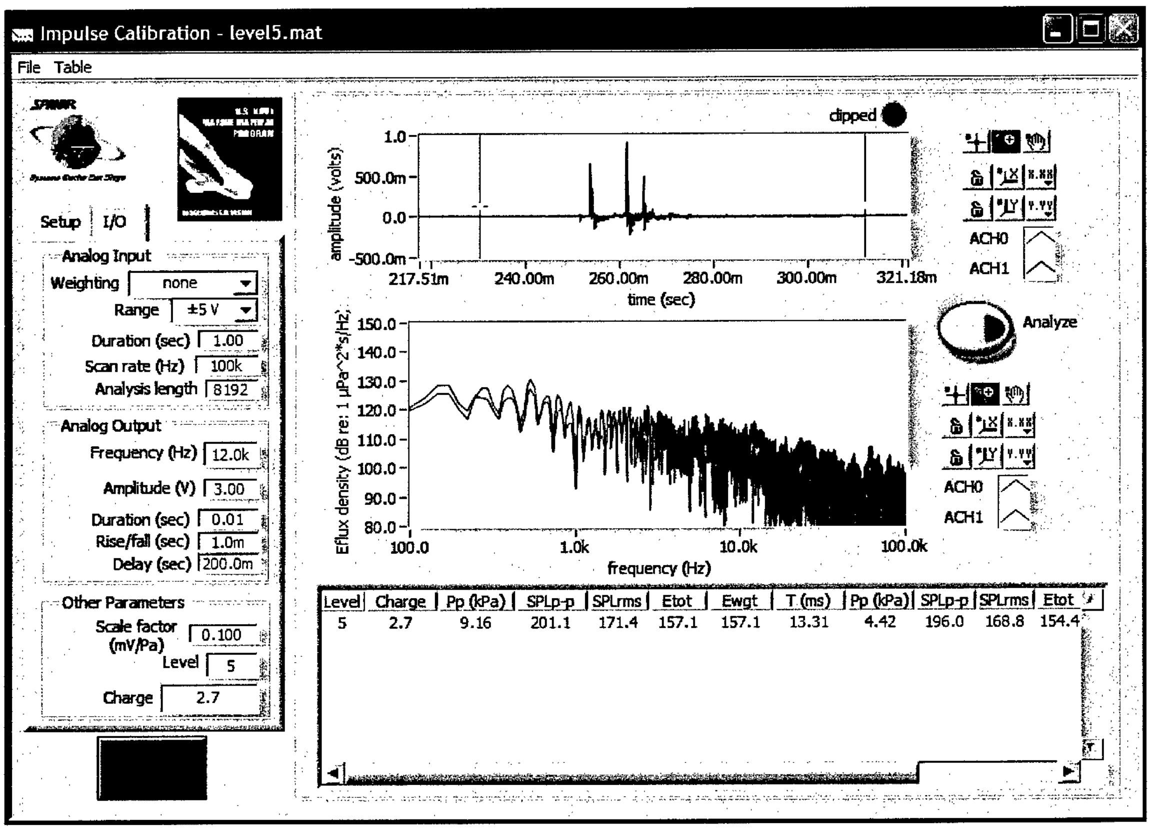


Figure 42. Impulse Calibration front panel.

12.2 PROCEDURE

Impulsive sounds are normally produced by external devices (such as a sparker or airgun) rather than by the DAQ boards residing within the PC. The PC triggers the impulsive source and digitizes the measured sound pressure. The PC may also be used to generate an S1 tone with some specific timing relationship to the impulsive source trigger. The impulsive source is triggered using the DAQ general-purpose counter output GPCTR0. This TTL-level digital signal triggers the hydrophone recording, so that the hydrophone recording begins when the impulsive source is triggered. Two hydrophones are normally used for impulsive sound measurements: one located at the estimated position of the subject's ears, the other located between the source and the first hydrophone. The relationship between the two hydrophone pressures is used to correct the measurement made (using the second hydrophone) during exposure of the subject to what would have existed at the location of the subject's ears (without the subject present).

To calibrate the impulsive source, first setup the source to the desired output parameters (e.g., airgun static pressure, sparker voltage). Next, check to make sure that the front panel **Scale Factor** value matches the charge amp settings. Set the front panel **Amplitude** to zero—this setting prevents an S1 tone from being produced during the measurement. Press the **Send S1** button to trigger the impulsive source and record the pressures measured by the hydrophones. The upper plot will show the measured time waveforms. Use the move cursor tool to move the left-most red line to a time value just before the impulsive waveform. Adjust the **Analysis length** (must be a power of two) to include the entire waveform of interest. Press the **Analyze** button to calculate the frequency spectra, amplitude parameters, and duration of the waveforms in the time range between the two red lines. The table shows the results for both hydrophones (ACH0 and ACH1) from left to right. Save the individual waveforms using the **File: Save As** command. Adjust the impulsive source parameters to achieve the desired pressure and/or energy values. Print the results using the **File: Print** menu command.

12.3 FRONT PANEL REFERENCE

Tables 50 and 51 list the Impulse Calibration front panel controls and indicators.

Table 50. Impulse Calibration **Setup** tab.

Option	Description
Devices and channels	
AI Device	Input device number for hydrophone signals.
ACH	Analog input channels to record.
Invert ACH0	Specifies whether the ACH0 waveform should be inverted (some hydrophones are wired in opposite polarity).
Invert ACH1	Specifies whether the ACH1 waveform should be inverted (some hydrophones are wired in opposite polarity).
AO Device	Output device number for S1 signal during calibration.
DAC	S1 signal output channel.
Max update rate	Update rate to use when generating the S1 signal.
Base path	Location of the HTP folder on the hard drive.

Table 51. Impulse Calibration **I/O** tab.

Option	Description
Analog Input	
Weighting	Specifies if a frequency weighting is to be applied to the measured frequency spectra. Options are **odontocete** and **pinniped**. These weighting networks are based on published measurements of hearing sensitivity in bottlenose dolphins (odontocete) and California sea lions (pinniped). A weighting network may give a better impression of an impulsive sound's impact on an animal by emphasizing frequencies within the audible range and lessening the importance of frequencies outside the audible range.
Range	Analog input DAQ range. Should be just larger than the maximum voltage produced by the hydrophone/charge amp.
Duration	Duration of hydrophone recording.
Scan rate	ADC conversion rate for hydrophone signals.
Analysis length	Number of scans to analyze. Must be a power of two.
Analog Output	
Frequency	S1 signal frequency.
Amplitude	S1 signal amplitude.
Duration	S1 signal duration (include rise and fall).
Rise/fall	S1 signal rise and fall times. A linear rise and fall is used.
Delay	Time delay between hydrophone recording start and generation of S1 signal.
Other Parameters	
Scale factor	Hydrophone/charge amp scale factor in mV/Pa.
Level	This text string is available for record-keeping purposes only. It has no effect on the measurements, but is included in the tabular results. The **Level** and **Charge** controls were intended to allow the operator to include two of the impulsive source parameters in the tabular results.
Charge	This text string is available for record-keeping purposes only. It has no effect on the measurements, but is included in the tabular results. The **Level** and **Charge** controls were intended to allow the operator to include two of the impulsive source parameters in the tabular results.

12.4 MENU REFERENCE

Tables 52 and 53 describe the Impulse Calibration menu options.

. Table 52. Impulse Calibration **File** menu.

Option	Description
Open	
Calibration Results	Opens a MAT file previously saved from the S1 and FS Impulse Calibration Save As menu. These files are MATLAB® binary files containing the measured time waveforms and frequency spectra.
S1 Recording	Opens an existing S1 file created during a hearing test session. These files are two-channel, interleaved, I16, Intel format with no header.
Save As	Creates a MATLAB® binary file containing the measured time waveforms and frequency spectra.
Print	Print the calibration results.
Close	Close this window and return to the HTP.EXE front panel.

Table 53. Impulse Calibration **Table** menu.

Option	Description
Save Table	Save the tabular results as an ASCII tab-delimited text file.
Clear All Rows	Erase all rows in the table.
Clear Selected Row(s)	Erase any selected rows in the table. Use shift-click to select more than one row.

12.5 DATA FILE FORMATS

12.5.1 MAT Files

MAT files are double-precision binary MATLAB® format files created by the MATLAB® save command and readable by the MATLAB® load command. They can be created on one machine and later read by MATLAB® on another machine with a different floating-point format, retaining as much accuracy and range as the disparate formats allow. Other programs, external to MATLAB®, can also manipulate them. The binary formats used vary depending on the size and type of any arrays. MAT files include variable names and numeric values.

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APPENDIX A: INTRODUCTION TO UNDERWATER SOUND

A.1 SOUND BASICS

A.1.1 Definition of Sound

Subjectively, the term *sound* refers to what is heard with the ears. Objectively, sound is a time-varying mechanical disturbance in an *elastic medium*—a medium which, when disturbed, will return to its equilibrium position after the disturbing influence is removed (Lindsay, 1988a). Sound cannot exist in a vacuum—an elastic medium such as air, water, steel, or even rock is required. In modern usage, *sound* refers not only to the phenomenon in air that one hears, but also to whatever else is governed by the same physical principles (Pierce, 1989).

Sound is produced when an elastic medium is set into motion by any means. Sound is most often created by a vibrating object within the medium. As the object moves back and forth, its motion is imparted to adjacent "particles" of the medium. The motion of these particles is transmitted to adjacent particles, and so on. The result is a mechanical disturbance (the "sound wave") that moves away from the source and propagates at a medium-dependent *sound speed.* As the sound wave travels through the medium, the individual particles of the medium oscillate about their equilibrium positions (they do not propagate with the sound wave). Because sound is an oscillatory disturbance that moves away from the source and transports no discernable amount of matter over large propagation distances, it is a *wave phenomenon* and sound "disturbances" are called sound waves.

As the particles of the medium move back and forth, they create small changes, or perturbations, about the equilibrium values of the medium density, pressure, and temperature. Sound at a particular point in space may be quantified by measuring these perturbations in the medium density, temperature, or pressure, or in some cases, by measuring the small movements of the particles of the medium. Sound waves in air and water are normally described using the pressure because it is easily measured.

Sound waves in fluids are *longitudinal waves*, that is, the particles of the fluid move parallel to the direction of the sound wave itself. Sound waves in solids may be longitudinal waves or *shear waves*; in shear waves, the particles move perpendicular to the wave direction. Fluids cannot support shear waves because the fluid simply "flows" instead. This document is primarily concerned with sound in fluids, particularly air and water.

A.1.2 Sound Speed

Sound waves in a particular medium travel at the medium *sound speed*, normally denoted by the variable c. The sound speed in a particular medium depends on the medium's compressibility and density—the more dense and "stiff" the medium, the faster sound travels.

The sound speed in a gas is given by $c = \sqrt{\gamma p_0 / \rho}$, where γ is the ratio of specific heats (for air, $\gamma = 1.4$), p_0 is the equilibrium (ambient) pressure, and ρ is the equilibrium (ambient) density. The sound speed in air is a function of the barometric pressure, temperature, and humidity; typical values are on the order of 340 m/s. The sound speed in a liquid is given by $c = \sqrt{B / \rho}$, where B is the bulk modulus (the bulk modulus relates the change in pressure produced by a change in density—it is a measure of the "stiffness" of a fluid). The sound speed in water is a function of temperature, depth, and salinity; a nominal value for freshwater or seawater is 1500 m/s. Sound speeds in freshwater are slightly lower than those for seawater at the same temperature and depth. Empirical formulas for sound speeds in water as functions of temperature, depth, and salinity may be found in Coppens (1981).

A.1.3 Physical Attributes of Sound

Sounds may be described in terms of physical and subjective attributes. Physical attributes are those that may be directly measured; subjective (or psychological) attributes may not be directly measured and require a listener to make a judgment about the sound. Subjective attributes include pitch and loudness. Physical attributes include amplitude, frequency, and phase.

To illustrate the physical attributes of sound, consider a pen and ink recorder connected to a pressure sensor located near a tuning fork and a drum, as shown in Figure A-1. If the tuning fork is properly struck, a sound wave will propagate past the sensor and the pen and ink recorder will trace out a graph of the time-varying pressure as shown in the upper graph. The pressure rises and falls about the equilibrium point (the atmospheric pressure) in a series of symmetric, equally spaced peaks and valleys. The type of sound produced by a tuning fork is an example of a type of sound called a "pure tone." For pure tones, the pressure, density, and temperature perturbations and the motions of the particles of the medium are sinusoidal functions of time; these types of sound waves are sometimes called harmonic waves or sinusoids. Harmonic waves rarely occur in everyday life, but are very useful because more complex sound scan be broken down into a sum of multiple harmonic waves using a mathematical theorem (Fourier's Theorem).

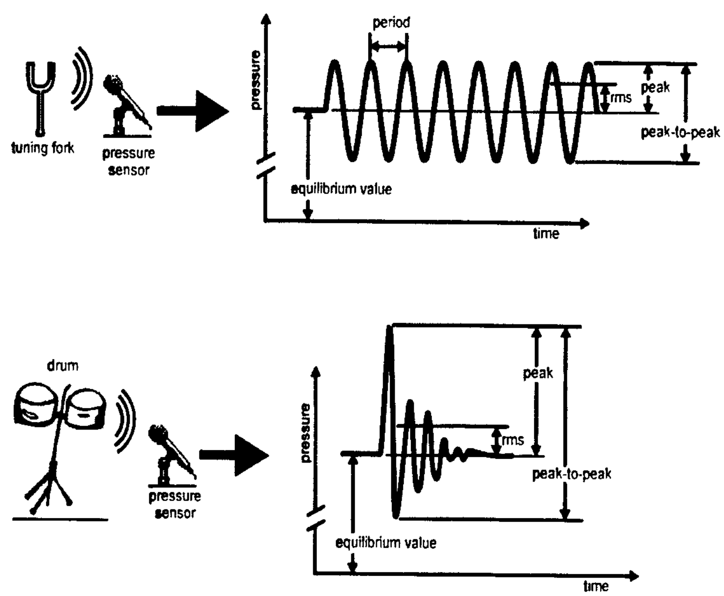

Figure A-1. Continuous-type and impulsive sound examples.

A single strike on the drum will also cause a sound wave to travel past the sensor; however, the graph produced by the pen and ink recorder reveals very different pressure fluctuations in this case. The pressure rises rapidly and then oscillates about the equilibrium position; the pressure rises and falls less with each additional oscillation, quickly returning to the equilibrium value. This type of sound is known as a transient, an "impulsive sound" or an "impulse."

A.1.3.1 Frequency

The number of times that the pressure (or another physical quantity used to quantify the sound wave) varies from its equilibrium value, through a complete cycle, in unit time is called the frequency. In one complete cycle, there is a positive variation from equilibrium, a return to equilibrium, then a negative variation, and a return back to equilibrium (Galloway, 1988a). Frequency is related to the speed at which the medium particles oscillate about their equilibrium positions. Frequency is the physical attribute most closely associated with the subjective attribute of *pitch*; the higher the frequency, the higher the pitch. The most common unit of frequency is the hertz, abbreviated "Hz"; 1 Hz is equivalent to one cycle per second. Mathematical operations involving frequency normally use the angular or circular frequency, $\omega = 2\pi f$, where f is the frequency in Hz. The unit of ω is radians per second (rad/s).

Pure tones have a constant, single frequency. Figure A-2(A) and (B) show two pure tones with different frequencies. Complex tones such as a note sounded on a guitar or piano contain acoustic energy at multiple, discrete frequencies rather than a single frequency. The relative contributions between the lowest frequency, called the fundamental, and the higher frequencies, called harmonics, determine the particular sound quality or timbre. Other sounds, such as white noise, which has a "hissing" sound, have energy spread out over a range of frequencies.

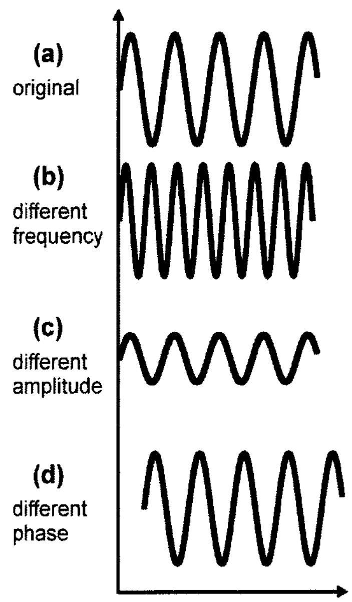

Figure A-2. (A) Comparison tone and tones with different (B) frequency, (C) amplitude, and (D) phase.

The human hearing range extends from approximately 20 Hz to 20 kHz. The note "middle-C" on a piano has a (fundamental) frequency of 261.63 Hz. Sound with a frequency above the highest frequency audible to humans is called *ultrasound*. Sound with a frequency below the human audible range is called *infrasound*.

A.1.3.2 Period

The time required for the pressure (or another physical quantity) to move through one complete cycle is called the *period*. The period is the reciprocal of the frequency, f (in Hz). The unit of period is the second.

A.1.3.3 Wavelength

The distance that a sound wave travels in one cycle is called the wavelength. The wavelength is usually denoted with the Greek symbol, lambda (λ). For pure tones, $\lambda = c/f$. The unit of wavelength is the meter.

A.1.3.4 Amplitude

The amount that the pressure (or another physical quantity used to quantify the sound wave) exceeds the equilibrium value indicates the sound *amplitude*. Amplitude is the physical attribute most closely associated with the subjective attribute, *loudness*. As the amplitude increases, the loudness also increases. The amount and physical units of the amplitude depend on the particular physical variable (i.e., pressure, density, temperature, or particle motion) used to describe the sound. Figure A-2 (A) and (C) show two pure tones with the same frequency but different amplitudes.

The *instantaneous amplitude* indicates the amount, at any instant of time, that the pressure (for example) exceeds the equilibrium value. The instantaneous amplitude is a time-varying quantity. The *peak amplitude* indicates the maximum amount that the instantaneous amplitude exceeds the equilibrium value. The *peak-to-peak amplitude* (p-p) indicates the difference between the maximum and minimum values of the instantaneous amplitude. For pure tones, the p-p amplitude is twice the peak amplitude. For other sounds with more complex instantaneous amplitudes, there may be no relationship between peak amplitude and p-p amplitude.

The *mean-squared* amplitude $\overline{A^2}$ is defined as

$$\overline{A^2} = \frac{1}{T}\int_0^T a^2(t)\,dt\,, \tag{3}$$

where $a(t)$ is the instantaneous amplitude and T is the time over which $a(t)$ is integrated (averaged). For sinusoids, it is common to integrate over an integer number of cycles; for other sounds, it is common to integrate over long time periods, that is to take the limit of equation (3) as $T \to \infty$. Since $\overline{A^2}$ does not have the same physical units as $a(t)$, it is common to use the *root-mean-square* (rms) amplitude instead. The rms amplitude, $\overline{A}$, is defined as the square root of the mean-squared amplitude:

$$\overline{A} = \sqrt{\frac{1}{T}\int_0^T a^2(t)\,dt}\,. \tag{4}$$

For pure tones (with T equal to an integer number of periods), equation (4) simplifies to $\overline{A} = A_p / \sqrt{2}$, where A_p is the peak amplitude. For more complex sounds, there is no fixed relationship between A_p and $\overline{A}$.

A.1.3.5 Phase

The phase of a sound wave indicates the portion of the cycle through which the sound wave has advanced, relative to some fixed reference time. For pure tones, the phase is often expressed in terms of degrees of angles, where one complete cycle contains 360° or 2π rad. For continuous sounds, the phase angle is only important for describing the relationship between multiple sound waves. Figure(A) and (D) show two pure tones with the same amplitude and frequency, but different phase angles.

A.1.4 Acoustic Signal Types

A.1.4.1 Tonal signals

Tonal signals (tones) may be described as sounds that may be decomposed into a finite number of individual pure tones. The spectrum of a tonal signal consists of one or more discrete frequencies. Tonal signals are normally described in terms of frequency, duration, and the rms amplitude.

A.1.4.2 Continuous-Type

A *continuous-type* sound is one that lasts for a relatively long time period, usually compared to the period for harmonic sounds. The higher the frequency of the sound, the shorter the duration may be for the sound to still be considered a continuous-type sound. A periodic sound with a duration of approximately 20 cycles or more may be considered continuous. Continuous-type sounds are sometimes referred to as continuous-wave (cw) sounds.

A.1.4.3 Transients and Impulses

A *transient* sound has a well-defined starting and ending time. Sounds produced by percussive events (e.g., striking a drum), or only a few cycles of an harmonic sound, are examples of transients. Technically, all sounds are transients; however, it is customary to consider sounds as transients only if their duration is short compared to some relevant time measure, often related to the frequency components within the signal (i.e., at lower frequencies, longer duration sounds may still be considered transients). An *impulsive sound* or *impulse* is a transient sound with a relatively short duration and a large amplitude (e.g., explosions or gunfire).

A.1.5 Time and Frequency Representations

A.1.5.1 Waveform

The plots of pressure versus time shown in Figure A-1 are examples of sound *waveforms*. The waveform is the instantaneous amplitude as a function of time. The waveform is an example of a *time domain* representation of an acoustic signal, because it portrays the variation in the sound amplitude as a function of time.

A.1.5.2 Frequency Spectra

An alternative to the time-domain representation of an acoustic signal is the *frequency domain* representation, where the signal amplitude is displayed as a function of frequency. This type of display is called a *frequency spectrum.*

Fourier's Theorem states that any periodic function may be expressed as a sum of simple harmonic terms (a Fourier Series) whose frequencies are integral multiples of the periodic function repetition rate. A periodic function is one whose instantaneous amplitude repeats itself after some time period (the period, T).

Fourier's Theorem may also be applied to aperiodic and transient sounds. For these cases, the sum of all harmonics is replaced by an integration over all frequencies. The specific mathematical operation is called a Fourier Transform (FT). It is common today to *sample* continuous time histories (such as the strip chart recordings shown in Figure A-1) to produce a sequence of digital values for later processing using a computer. The Fast Fourier Transform (FFT) is an efficient numerical algorithm used to compute the Fourier Transform of a digital signal.

A.1.5.3 Spectrogram

A sound that changes in time, such as a spoken word or a marine mammal vocalization, can be more completely described by examining how the frequency spectrum changes with time. In a graph called the sound *spectrogram*, the frequencies of the complex sound are plotted versus time, with the more intense frequency components shown in the third dimension or as a lighter or colored point on a two-dimensional graph. Since pure tones have a constant frequency, they appear as horizontal lines on the spectrogram.

A.1.6 Pressure and Particle Motion

A.1.6.1 Acoustic Pressure

Pressure is defined as the amount of force per unit area. The *acoustic pressure* (or sound pressure) is defined as the incremental variation in the static (equilibrium) pressure within a medium as a sound wave travels through it (Galloway, 1988b). Acoustic pressure thus indicates the perturbation of the medium pressure from its equilibrium value. The unit of pressure is the Pa. In older literature, acoustic pressures were often presented in units of microbar (μbar); 1 μbar = 1 dyne/cm^2 = 0.1 Pa. Acoustic pressures are also sometimes specified in units of pounds per square inch (psi); 1 psi ≈ 6890 Pa. Normal atmospheric pressure is approximately 10^5 Pa. The lowest sound pressures audible to humans are on the order of 20 μPa (rms).

A.1.6.2 Acoustic Particle Motion

Sound waves may also be described using the acoustic particle motion, that is, the oscillatory displacement, velocity, or acceleration of the actual "particles" of the medium. Particle motions should not be confused with the motion of the sound wave itself, or with any "bulk" motion of the medium itself, such as air or water currents. The preferred units of particle motion are the customary units for displacement, velocity, and acceleration: m, m/s, and m/s^2, respectively. Particle displacements associated with typical music and speech sounds are only a small fraction of a mm. At the lower limits of human hearing, acoustic particle displacements in air are roughly on the order of 10^{-12} m.

Particle motions are created by spatial gradients in the pressure: the pressure gradient creates a force that acts upon the mass of the medium to cause it to move (Newton's Second Law of Motion).

Direct measurement of particle motion is difficult and practical only at low frequencies. It is more common to estimate particle motion from multiple pressure measurements taken over some spatial region.

Particle motion is a vector quantity—it has magnitude and *direction.* In contrast, pressure, temperature, and density are scalar quantities, and possess magnitude only.

A.1.7 Plane and Spherical Sound Waves

A.1.7.1 Plane Waves

If the acoustic quantities associated with a sound wave vary with time and a single spatial coordinate, but are independent of position within planes normal to this coordinate, the sound wave is called a *plane wave*. A simple example of plane waves would be sound produced at low frequencies inside a long, uniform tube with rigid walls. In this case, the acoustic quantities vary along the length of the tube; however, at any axial position, the acoustic variables are uniform across the tube diameter. If the sound field is unbounded so that the sound waves may propagate outward from the source free from any reflective surfaces, the sound wave is called a *plane progressive wave*. The pressure in a plane wave is a sinusoidal function of time and distance. The peak amplitude is constant. Although an idealization, the concept of plane progressive waves is useful when analyzing real sound waves, in part, because sound waves from any source approach plane wave behavior at large distances from the source.

A.1.7.2 Spherical Waves

In addition to the plane wave, another common idealization of an acoustic disturbance is a spherically symmetric wave spreading out from a source in an unbounded medium. In contrast to the plane wave, the amplitude of a spherical wave decreases with distance from the source. For each doubling of the radial distance from the source, the amplitude is halved.

A.1.8 Sound Levels and the Decibel

A.1.8.1 Logarithmic Units and the Origins of the Decibel

In part because of the very large dynamic range of the human auditory system (the largest and smallest amplitude sounds that may be heard vary from roughly 10^{-6} Pa to 60 Pa), sound amplitudes are often expressed as *sound levels*. The term "level" indicates the logarithm of the ratio of a given quantity to a reference quantity with the same units (Young, 1988). The use of a logarithmic scale compresses the range of numerical values that must be used. Another motivation to describe sounds with a logarithmic scale is the fact that humans judge the relative loudness of two sounds by the ratio of their pressures, which is a logarithmic behavior (Kinsler *et al.*, 1982).

When using logarithmic units, the base of the logarithm and the reference value must be specified. Typically, the logarithm is taken to the base 10 (so the logarithm is written as $\log_{10}$). The logarithm of a number, y, to a base, b, is the power of b that equals y: if $x = \log_{10} y$, then $y = 10^x$. As an example, $\log_{10}(100) = 2$, since $10^2 = 100$. Some important mathematical relations involving logarithms are listed in equation (5):

$$\begin{aligned}\log_b(xy) &= \log_b x + \log_b y \\ \log_b(x/y) &= \log_b x - \log_b y \\ \log_b x^a &= a \log_b x\end{aligned} \tag{5}$$

Sound levels are normally expressed in *decibels*. A decibel is 1/10 of a bel, a logarithmic unit named after Alexander Graham Bell. To express a quantity, X, in decibels using a reference, X_{ref}, the equation is

$$10\log_{10}\left(\frac{X}{X_{ref}}\right), \tag{6}$$

if X and X_{ref} have units of power or energy (including intensity), or

$$20\log_{10}\left(\frac{X}{X_{ref}}\right) = 10\log_{10}\left(\frac{X^2}{X_{ref}^2}\right), \tag{7}$$

if X and X_{ref} have units of pressure, velocity, voltage, or a similar quantity. The use of X^2 and X_{ref}^2 arises from the fact that power and energy are related to the product of pressure and velocity or to pressure squared or velocity squared.

A.1.8.2 Reference Values

When a numeric value is presented in decibels, it is crucial that the numeric value and units of the reference quantity are specified. Decibel values should also include the specific amplitude measurement, i.e., peak, rms, etc., if there may be any ambiguity. The accepted terminology for specifying decibel quantities is to present the numeric value, followed by the text "re" and the numeric value and units of the reference quantity. For example, a pressure of 1 Pa, expressed in decibels with a reference of 1 μPa, would be written as 120 dB re 1 μPa. Table A-1 lists the recommended reference quantities for acoustic pressure and acoustic intensity in air and underwater.

Table A-1. Preferred decibel references.

Quantity	Reference
pressure (air)	20 μPa
pressure (water)	1 μPa
intensity	10^{-12} W/m^2

The in-air pressure reference of 20 μPa and the intensity reference of 10^{-12} W/m^2 were chosen because they correspond to the approximate pressure and acoustic intensity in air at the limits of human hearing. In older literature, a reference pressure of 1 μbar was often used for underwater sound. To convert from dB re 1 μbar to dB re 1 μPa, add 100 dB: 50 dB re 1 μbar = 150 dB re 1 μPa.

The exception to the rule regarding specifying reference quantities is when one is referring to a *difference* in decibels. Subtracting two quantities expressed in decibels (with the same reference quantity) is equivalent to taking the ratio of the original two quantities; thus, there is no reference quantity.

A.1.8.3 Sound Pressure Level

Perhaps the most common decibel quantity encountered in acoustics is the sound pressure level (*SPL*). The *SPL* is defined as

$$SPL = 20\log_{10}\left(\frac{\overline{P}}{P_{ref}}\right), \tag{8}$$

where the recommended reference pressure, $P_{ref,}$ is listed in Table A-1 for air or water. Note that the *SPL* is defined using the rms pressure.

For transient signals, the peak or p-p amplitude is often used in place of the rms amplitude in *SPL* calculations. In these situations, some indication should be made that the *SPL* values are not rms measures.

A.1.8.4 Source Level

The term *source level* (*SL*) refers to the sound amplitude produced by a source at some reference distance from the source, typically 1 m. Often, the *SL* is reported in pressure units; thus, an *SL* may be indicated as, for example, 150 dB re 1 μPa at 1 m, or 150 dB re 1 μPa-m.

A.1.8.5 Received Level

The term *received level* (*RL*) refers to the sound amplitude heard by a listener or measured by a sensor at some distance from the source rather than the amplitude produced at the source itself. Normally, the *RL* is reported in pressure units; thus, an *RL* may be indicated as, for example, 120 dB re 1 μPa.

A.1.9 Ambient Noise

Ambient noise may be described as the unwanted sound arising from unidentified sources within the medium. Sources of airborne ambient noise include winds, traffic, aircraft, and birds. Underwater ambient noise is created by waves, wind, ocean currents, shipping, and biological sources (e.g., snapping shrimp and marine mammals).

A.2 SOUND PROPAGATION

A.2.1 Reflection and Refraction

When a sound wave propagating in a medium encounters a second medium with a different characteristic impedance, part of the incident sound will be *reflected* back into the first medium and part will be *transmitted* into the second medium. If the second medium has a different sound speed than the first, the propagation direction will change as the sound wave enters the second medium; this phenomenon is called *refraction*. Refraction may occur not only at the boundary between two different media (e.g., air-water), but within a single medium if spatial gradients exist in the sound speed. The relative amplitudes of the reflected and transmitted sounds depend on the densities and sound speeds of the two media and the angle at which the sound is incident upon the boundary between the two media.

A.2.2 Scattering, Diffraction, and Reverberation

Sound waves experience *diffraction* in much the same manner as light waves. Diffraction may be thought of as the bending of a sound wave around an obstacle. Common examples include sound heard from a source around the corner of a building and sound propagating through a small gap in an otherwise closed door or window. Through diffraction, new sound waves are created at the edges of obstacles in the sound path. These new sound waves may spread into the geometrical "shadow" created by the incident wave (see Figure A-3) and also interfere with the original, incident sound. The effects of diffraction become less pronounced as the sound wavelength decreases relative to the size of the diffracting obstacles; with ultrasonic sound sources, it is possible to produce sound shadows (Lindsay, 1988a).

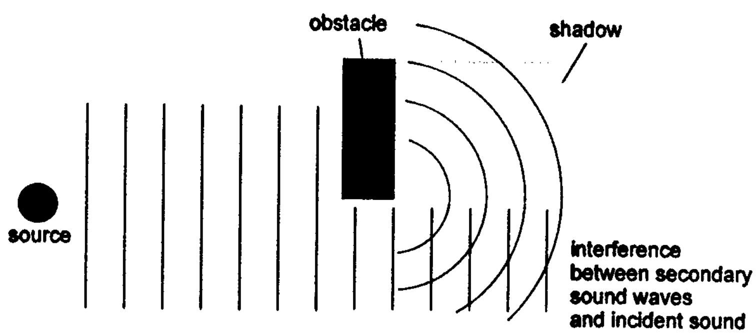

Figure A-3. Acoustic "shadow" created by sound diffracting around an object.

An obstacle or inhomogeneity in the path of a sound wave causes *scattering* if secondary sound spreads out from it in various directions (Pierce, 1989). Scattering is very similar to diffraction; normally, the term diffraction describes sound bending or scattering from a single object and the term scattering is used when there are multiple objects. Common inhomogeneities that cause scattering include suspended particles, smoke, fog, suspended gas bubbles, and fish. Lower frequency sounds typically scatter less than higher frequency sounds.

Reverberation refers to the prolongation of a sound that occurs when sound waves in an enclosed space are repeatedly reflected from the boundaries defining the space, even after the source has stopped emitting.

Scattering, diffraction, reverberation, and reflection all share the same basic physical mechanisms. If the boundaries are large and the surface roughness small compared to a wavelength, the process is normally viewed as reflection. If surfaces are rough or irregular, though large, the incident sound is considered to have suffered boundary scattering; volume scattering refers to scattering from a unit volume of the media (Maples, 1988).

A.2.3 Sound Attenuation

As a sound wave passes through a medium, the intensity decreases with distance from the sound source; this process is known as *attenuation*, *transmission loss*, or *propagation loss*. The main contributors to sound attenuation are (1) geometrical spreading of the sound wave as it propagates away from the source, (2) sound absorption (conversion of sound energy into heat), (3) scattering, (4) diffraction, (5) interaction with boundaries, and (6) multipath interference.

A.2.3.1 Spreading Loss

Spreading loss is a geometrical effect representing a regular weakening of a sound wave as it spreads out from a source; spreading describes the reduction in sound pressure and intensity caused by the increase in surface area as the distance from a sound source increases. Spherical and cylindrical spreading are the most common types of spreading loss.

A point sound source in a homogeneous, lossless medium without boundaries will radiate spherical waves; the acoustic energy spreads out from the source in the form of a spherical shell. As the distance from the source increase, the shell surface area increases. Assuming a fixed acoustic power, the average intensity must therefore decrease with distance from the source (average intensity is power per unit area). The surface area of a sphere is $4\pi r^2$, where r is the sphere radius; thus, the change in intensity is proportional to the radius squared and the pressure decreases as the inverse of radial distance. This prediction is known as the *spherical spreading law* and is equivalent to a 6-dB reduction in pressure for each doubling of distance.

When a medium has plane parallel upper and lower boundaries, sound spreading is not spherical because sound cannot cross the boundaries. In *cylindrical spreading*, spherical waves expanding from the source are constrained by upper and lower boundaries and thus approach a cylindrical shape; the wave front expands in the shape of a cylinder rather than a sphere. Cylindrical spreading is an approximation to wave propagation in a water-filled channel with horizontal dimensions much larger than the depth. Cylindrical spreading predicts a 3-dB reduction in sound pressure for each doubling of distance from the source.

A.2.3.2 Absorption Loss

Sound absorption is the conversion of sound energy into heat energy. A sound wave traveling through any medium will lose a certain fraction of its intensity in each unit distance traveled. The primary causes of sound absorption in air and water are viscosity, heat conduction, and molecular relaxation processes (Lindsay, 1988b; Kinlser *et al.*, 1982; Urick, 1983).

A.2.3.3 Scattering Loss

Scattering loss results when inhomogeneities within the medium scatter sound energy away from the direction in which the sound wave is traveling, so that the amplitude of the sound wave decreases.

A.2.3.4 Diffraction Loss

Diffraction loss results when sound diffracts around an obstacle in the sound path; because sound energy tends to "leak" into the shadow created by the obstacle, energy will be lost from the original sound wave.

A.2.3.5 Boundary Loss

When a sound wave encounters a second medium (e.g., the ground for airborne sound or the ocean bottom or surface for underwater sound), some of the incident sound energy will be reflected and scattered and some will be transmitted into the second medium, resulting in loss of acoustic energy compared to the original sound wave.

A.2.3.6 Multipath Loss

The term *multipath* refers to sound waves from a single source traveling multiple sound paths before reaching a single receiver. Multipath propagation is common when a source is located relatively close to a boundary and, in underwater applications, when the depth is small relative to the propagation distance. In multipath propagation, sound may not only travel a direct path from source to receiver, but may also be reflected from the surface and/or bottom multiple times before reaching the receiver. The existence of multipaths results in a condition that permits constructive and destructive interference between sound waves propagating in the different paths and the received sound amplitude may be reduced as a result.

A.2.4 Underwater Sound Propagation

A.2.4.1 Refraction

Refraction of sound resulting from spatial variations in the sound speed is one of the most important phenomena that affect sound propagation in water. The sound speed in the ocean is primarily a function of hydrostatic pressure (i.e., depth) and temperature; sound speed increases with hydrostatic pressure and temperature. In seawater, temperature has the most important effect on sound speed for depths less than about 300 m. Below 1500 m, the hydrostatic pressure is the dominant factor because the water temperature is relatively constant. The variation of sound speed with depth in the ocean is called a *sound speed profile.*

A.2.4.2 Sea Surface Effects

Because it reflects and scatters sound, the sea surface has a major effect on the propagation of underwater sound in applications where either the source or receiver are at shallow depth. If the sea surface is smooth, the reflected sound pressure is nearly equal to the incident sound pressure; however, if the sea surface is rough, the amplitude of the reflected sound wave will be reduced. The motion of the sea surface also affects the frequency of the reflected or scattered sound waves by adding "sidebands" about the incident sound spectrum. The sidebands are duplicates of the spectrum of the surface motion (Urick, 1983). The motion of the surface, therefore, widens or "smears" the spectrum of an incident sound wave with a narrow frequency bandwidth.

For a particular sound source, the phase relationship between the "direct" sound wave, which propagates directly from the source to the receiver, and the reflected wave depends on the depth of the source and the distance to the receiver. At some distances, the reflected wave will be in-phase with the direct wave (their waveforms add together), and at other distances, the two waves will be out-of-phase (their waveforms cancel). This situation results in constructive and destructive interference between the surface reflected sound wave and produces an interference pattern in the underwater sound field. This phenomenon is called the *Lloyd mirror* effect and is an example of multipath propagation loss. In this case, the resulting sound field contains an alternating series of pressure maxima and minima. For a perfectly smooth surface, the pressure minima extend to zero; for rougher surfaces, the minima do not extend to zero (there is imperfect cancellation).

A.2.4.3 Sea Bottom Effects

The sea bottom is a reflecting and scattering surface, similar to the sea surface. Sound interaction with the sea bottom is more complex, however, primarily because the acoustic properties of the sea bottom are more variable and the bottom is often layered into regions of differing density and sound speed. The Lloyd mirror effect may also be observed from sound sources located near the sea bottom. For a "hard" bottom such as rock, the reflected wave will be approximately in-phase with the incident wave. Thus, near the ocean bottom, the incident and reflected sound pressures may add together, resulting in an increased sound pressure near the sea bottom.

A.2.4.4 Shallow-Water Acoustics

Sound propagation in shallow water is complicated because of sound interaction with the surface and bottom, as well as sound speed variations within the water column. Shallow water sound fields are highly variable and site-specific (Urick, 1983; Richardson *et al.*, 1995). It is therefore difficult to develop theoretical models to predict sound propagation in shallow water. Theoretical analysis are often comb ined with site-specific empirical data to obtain reliable predictions (Richardson *et al.*, 1995).

A.3 ACOUSTIC MEASUREMENTS

A.3.1 Acoustic Receivers

A.3.1.1 Microphones

A transducer is a device that converts energy from one form to another. A *microphone* is a transducer that converts airborne sound into an electronic voltage. Most microphones are pressure sensors; they convert pressure fluctuations into a proportional voltage. Important microphone parameters include the sensitivity (ratio of output voltage to applied pressure), useable frequency range, dynamic range (range of loudest to quietest sounds that may be measured), equivalent noise level (sound pressure required to produce an output voltage equivalent to the microphone's self-noise), and directionality. The most common microphone types are electrostatic (condenser), piezoelectric, dynamic, magnetic, and carbon (Beranek, 1988).

A.3.1.2 Hydrophones

A hydrophone is an electroacoustic transducer that converts underwater sound into an electronic voltage. Most hydrophones measure acoustic pressure. The majority of hydrophones are piezoelectric transducers; a piezoelectric device is one which produces an electric charge in response to mechanical deformation produced by an applied stress (pressure). Piezoelectric hydrophones thus produce an electric charge proportional to the applied acoustic pressure. Important hydrophone parameters include the sensitivity, useable frequency range, dynamic range, equivalent noise level, directionality, and maximum operating depth.

A.3.2 Acoustic Sources

A.3.2.1 Projectors

Sound projectors are electroacoustic transducers that convert electric voltage or current into sound waves. For applications in air, dynamic (moving-coil) loudspeakers are the most common type of sound projector. Underwater sound projectors include dynamic types (e.g., USRD J-series transducers) or piezoelectric. Underwater sound projectors that require high-power output (such as certain sonar sources) often use a flextensional design where a "stack" of piezoelectric material is used to deform a surrounding elliptical shell. Sound projectors are normally used to produce tonal or continuous-type noise; it is very difficult to produce short-duration transients or impulses with sound projectors—when excited with an impulsive voltage, the projector will tend to "ring" at its resonant frequency, producing oscillations in the pressure waveform. Important parameters for sound projectors include the transmitting response (amount of sound pressure produced per unit applied voltage or current), frequency response, resonant frequency, maximum input power, directionality, and maximum operating depth (for underwater applications).

A.3.2.2 Explosions

Chemical explosives produce a volume of rapidly expanding gas in an extremely brief period. Chemical explosives are of two types: (1) detonating, or "high," explosives, and (2) deflagrating, or low, explosives. Detonating explosives, such as TNT and dynamite, are characterized by extremely rapid decomposition and development of high pressure, while deflagrating explosives, such as black and smokeless powders, are merely fast burning and produce relatively low pressures with a slower rise time.

In air, explosions produce an impulsive pressure wave that is heard as a loud "crack" similar to the sound of thunder. The pressure waveform produced by an explosion in air resembles an ideal impulse: a rapid rise to a peak pressure, followed by a return to the equilibrium pressure (zero acoustic pressure). In practice, reflections from the ground or other objects tend to lengthen and distort the received pressure waveform at a given location. Pressure signatures measured from in-air explosions typically contain an initial high-pressure peak followed by multiple, smaller amplitude peaks caused from multipath propagation and reflections.

In underwater applications, high explosives like TNT and its derivatives (such as HBX-1), produce a spherically symmetric shock wave along with (in sufficiently deep water) an oscillating globular mass of gaseous materials (Weston, 1960; Urick, 1967). At close range, the pressure signature of an underwater detonation of a high explosive in deep water therefore consists of the shock wave followed by a number of bubble pulses (Urick, 1967). At longer ranges, the signature is complicated by surface and bottom reflections, attenuation, and refraction effects. At shallower detonation depths, the mass of gaseous material may be blown in to the air, thus reducing or eliminating the bubble pulses. Pressure signatures of underwater explosions at large distances therefore cannot simply be produced by smaller charges at close range. Peak SPLs of underwater explosions may exceed 250 to 260 dB re 1 μPa -m (Richardson *et al.*, 1995).

A.3.2.3 Airguns

The airgun is a pneumatic sound source designed to release a specified volume of high-pressure air into the water. The rapid release of air produces a shock wave followed by several oscillations resulting from the repeated collapse and expansion of the air bubble. Airguns produce impulsive waveforms with mainly low-frequency content. Airguns are mainly used as underwater sound sources for seismic profiling—that is, "imaging" of the ocean bottom and subsurface by measuring the amplitude and time delay of sound pulses reflected from the ocean bottom. Air guns used in geophysical surveys are routinely operated at peak source levels exceeding 210 dB *re* 1 μPa-m (Richardson *et al.*, 1995).

A.3.2.4 Waterguns

A *watergun* is another pneumatic sound source used in geophysical surveys. The watergun is divided into two chambers: an upper firing chamber, which contains compressed air, and a lower chamber, which is filled with water. When the gun is fired, the compressed air forces a shuttle downward and expels the water from the lower chamber. The shot of water leaving the gun creates a void behind it; the collapse of water into this void creates an acoustic wave. Waterguns produce impulsive pressure waves with generally higher frequency content than airguns.

A.3.2.5 Sparker (Arc-Gap Transducer)

In a *sparker*, a large electrical voltage is generated between two oppositely charged terminals underwater. The terminals are then brought incrementally closer until an electrical arc bridges the small gap between them. The electrical arc momentarily vaporizes the water between the terminals, producing gas bubbles that quickly collapse. The collapsing bubbles produce an impulsive sound.

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14. ABSTRACT

This document describes the operation of the computer program HTP.EXE, which was designed to allow computer control of marine mammal hearing tests. The system uses the psychophysical "staircase" method and a behavioral response paradigm to estimate hearing thresholds in trained subjects. The program is suitable for isolated hearing threshold measurements and paired measurements made before and after exposure to threshold-affecting treatments (e.g., exposure to intense sound or other ototoxic agent). Although specifically designed for use with marine mammals, the program is appropriate for use with any subject trained for participation in a behavioral response paradigm.

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